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August 2003

Home Recording ™

THE HOW-TO MAGAZINE FOR MUSICIANS

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Deep & Wide: Ambient Recording

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RECORDING AMPLIFIER

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60 Hz
1 A

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LIFT
GROUND

MODERN   CH1-CH2   SOLO

EXT. TRIGGER
TO POWER AMP   EXTERNAL SWITCHES

LIVE OUT

TO PWR AMP
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MESA ENGINEERING

CLEAN   BRIGHT
FAT
BRIT

INPUT
FT. SW.

GAIN   -3db   TREBLE   MID   BASS   PRESENCE

RAW
VINT
MODERN

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WIN the Samson/Zoom Giveaway! Page 77

Gear

AXE HANDLE

Guitar is a uniquely expressive tool—and therefore resistant to the easy cloning of more percussive instruments.

Last issue, our cover story examined ways to handle bass frequencies in recording. This issue, we move up the frequency spectrum into the territory occupied by electric and acoustic guitars.

The interesting thing about guitar is that, as an ad exec might say, it's often imitated but never duplicated—by samplers, anyway. The fact that the guitar, like the saxophone, is a uniquely expressive tool—and therefore resistant to the easy cloning of more percussive instruments—makes capturing it on disc or tape a unique challenge. *HR* is lucky to have recruited several guitarist/engineer/writers to provide you with excellent advice on recording your axe. Check out their tips beginning on page 32.

Our series "The Killer Apps" also marches on, and now it's Logic Audio's turn to shine. New contributor Joe Albano, a studio consultant and longtime Logic user, points out the pluses that have made this newly Mac-only app one of the fastest growing programs in terms of popularity with new users.

One of the hottest R&B/funk/synth bands of the '80s was the System, led by vocalist Mic Murphy and keyboardist/producer David Frank. The band was widely admired by musicians for the deft touch it brought to songwriting and production. David Frank has carried on as a major behind-the-scenes force for several current R&B-oriented pop acts. In this month's ProView Howard Massey heads to L.A. to get Frank's views on sounds and songs.

Also, pro studio owner Rich Salz opens our minds and ears to the concept of ambient recording, the technique of using spatial elements to give your recordings a width and depth often difficult to achieve. Proper use of ambience can separate your work from the run-of-the-mill. As our friend the ad exec might say, "You can't touch it! You can't feel it! You can't smell it! But it's there working for you 'round the clock!" And you can certainly hear it, although figuring out how it's done is more difficult. That's where Rich comes in. Check it out on page 50.

Finally, take note of David Darlington's review of the Roland VariOS. We love this concept: an external hardware module that's controlled by software in the computer and into which you can download several of Roland's classic synths for—get this—free. It's an ongoing trend that blends the best of both worlds—the power of external hardware and the flexibility of software. We see this trend with other units like TC Electronic's PowerCore (now available in a FireWire rackmount unit). Watch for more helpful hardware like this—and more *HR* coverage of this new paradigm—in the near future.

Rusty Cutchin

The latest blow to the record industry.

SONAR STRATEGY

Regarding Thad Brown's response to Kevin Williams' question about getting the loudest final mixdown in Cakewalk Sonar [Q&A, JUN/03], I would like to contribute my own trial-and-error–honed method. I have been working almost exclusively in Sonar for more than a year and have developed a step-by-step process for my mixdowns that seems to work pretty well. When I say step-by-step, I really mean it, because improper gain setting any time during this process will result in that nasty clip distortion that is so problematic within the domain of digital recording (at least for those of us that cannot yet afford the best front-end compressors and audio-to-digital converters).

Tracking: Thad's comment on starting with the loudest possible tracks is important, but it should be noted that any track recorded with clipping cannot be saved. If the clip is on the recording, it is there to stay. It is better to record a relatively quiet track if the highest volume peaks do not exceed the limit of your A/D converter or Sonar. Sonar provides a very handy red peak indicator on the meters that stays on if a track has clipped. When adjusting your input trim levels prior to recording, pay close attention to this light. Make sure the source being recorded hits the loudest possible note as you adjust the inputs so that the meters barely touch the red. It is even a good idea to set the trim a little lower; most of us (especially those damn guitar players) will play or sing a great deal louder once the music starts. Immediately after laying the track, check to see if any of those red peak indicators have turned on. If they have, then you have already recorded a clip. You can also see any clips on the wave graphic as any line that exceeds the boundary of the track. If you are recording a source with extreme transients, you may want to utilize an outboard compressor in front of the A/D converter.

Mixdown: During mixdown, first do a prelim mix, adding plug-in effects and automations accordingly. When you are happy with each individual track and the global balance, play back the entire song while paying close attention to the meters. (It is okay if the meters peak out on this playback; you will fix them in a minute.) Identify the hottest track and adjust the trims (in Sonar) so that the red meter lights do not quite come on for that track. Then adjust the rest of the mix to a proper balance, being careful not to alter the volume setting that you have already established for that first hot track. When you are done, the hot track should still be the loudest overall. If you cannot get certain parts of an individual track to sound loud enough without the red peak indicator lighting up, try using plug-in compressors to limit the transients. You can even create a volume track envelope. Find the spot that causes the clip, and automate a volume adjustment for an individual peak. (This is also good for removing plosives on vocal tracks.)

Mastering: After bouncing your mixdown to two tracks, save it as a new file. (I always add the suffix "2T" to the file names to indicate that it is a mastering session.) Remove all the other tracks, and listen to the newly created stereo mixdown track. Once again, if those red lights on the meters light up, you already have clipping. (You must go back to step 2 and try again.) If not, go ahead and add your mastering effects. Thad mentions several mastering tools, like T-Racks. I have been using Isotope Ozone, with mixed (sometimes awesome) results. Before exporting the audio, listen to the whole track very carefully (you are almost done), and turn the volumes up until you find the point that is as loud as it can be without causing that little red peak indicator to ruin your day.

Finally: There are many different stages of the process in which clip distortion can be introduced to your Sonar project. By paying careful attention to your meters and especially the peak indicators at each step, you can ensure that you achieve the loudest possible final master without ruining it with those terribly un-euphonic cracks and snaps. But remember, your final masters will probably never sound quite perfect to you; in fact, they shouldn't. The production process involves many tradeoffs and compromises along the way. Your job as producer is to find the balance between everything, where all of the tracks sound as good as they possibly can together. Best of luck.

—*Matthew E. Guski*

Matthew, this is all great advice, and you are right that one of the first commandments of digital audio is Thou Shalt Not Clip the Audio Hardware While Tracking. Keeping an eye on the Sonar clip lights and using an outboard compressor on the way to the sound card are good ways to avoid this. Your method of mixing—watching the hottest track and using that as a reference—is also one solid way to keep digital levels where they should be when working on a track.

—*Thad Brown*
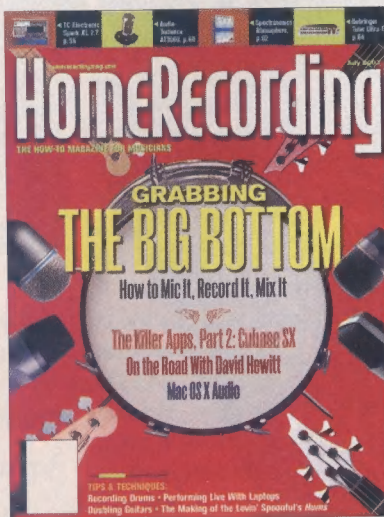
Send your questions and input to **Feedback, Home Recording Magazine, 6 E. 32nd St., 11th FL, New York, NY 10016,** *or reach us by e-mail at homerecording@cherrylane.com. Check out our Web site at* homerecordingmag.com.

Lean, Mean, Mixing Machine

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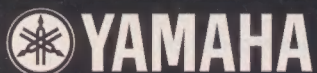
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MINI ISSUES TO DEAL WITH

To Thad Brown: I'm a hardware/software geek (recording and performing) musician, as you are. I am very interested in the mini PCs, especially as they're being touted on *tomshardware.com*. I really like the heat pipe technology and incredible "transportability," but I'm concerned about the reported heat buildup from two internal hard drives. (After all, one drive should be for the OS and the other for data, right?) Also, does the Shuttle config (with the wonderfully quiet heat pipe cooling mechanism) really allow for the necessary expansion that a modern project studio would need?

And just so you and all the others at *HR* know, I think you have a great home-recordist market focus, and I have recommended this mag as a must-read to quite a few friends and associates.

Cheers, and keep up the fantastic work.

P.S. Thanks for the hot-rod mic article [MAY/03]. I bought my Marshall MXL 2001 mic for the upgrade and will be purchasing the Mojave Audio upgrade soon. I can't wait, and I credit your mag exclusively with making this wonderful hardware hack opportunity known to me. I am communicating with Mojave Audio about their upgrade and will provide them with full feedback and continued kudos to *HR*.

— *Brian D.*

Brian, it sounds like we visit the same Web sites. The mini PCs are really exciting computers, and I've given serious thought to buying one myself, but I just don't think they are the right option for audio recordists yet. You're smart to be worried about computer noise, but don't think that just having a heat pipe instead of a traditional air cooler means something is quiet. For example, in the last tomshardware.com review of mini PCs, the Shuttle model (with a heat pipe) still averaged 50 dB(A). The site didn't say how or from what distance this was measured, but noted that the testers in the lab complained about the noise. No matter how it's sliced, 50 dB is way too loud for an audio computer unless there is no other option.

The second potential problem is expandability. Mini PCs have at most one PCI slot; some have none. Most audio users will need that PCI slot for an audio card. At absolute minimum, I need two PCI slots—one for my RME audio card and one for my UAD-1, so that kills the mini PC for me right there. If someone is absolutely sure that there will never be a need for more than one PCI card or that a FireWire audio interface will get the job done, the mini PC might be an option.

The final issue is cost. For kicks, I just spec'd out an nForce2-based mini PC system from Shuttle. After I add 512 MB of memory, a processor, a graphics card, a 40 GB hard drive, and a cheap CD-R/RW, the price gets close to $800, and that's without a keyboard, mouse, operating system, or display. For a little under $1,200, I can get a similarly performing Dell laptop that is quieter, can be used almost anywhere, and includes the keyboard and display. It has some of the expansion problems of a mini PC, but a true portable also has significant benefits.

This isn't to say that a mini PC is never the right option; I think the general trend toward smaller, less intrusive computers will continue, and at some point we'll probably all buy them. I'm just not sold quite yet.

— *Thad Brown*

SOFTWARE ON THE CHEAP

I bought the JUN/03 issue of *Home Recording* and couldn't find an opinion on computer software for those who don't want to spend more than $100 at a time on a recording software application. I have considered Cubase VST and Home Studio by Cakewalk, but haven't found anybody who can put into words why I should go with one or the other. Plus, there must be others out there. For example, I am downloading Pro Tools Free as we speak. I have a PowerMac G4 350 MHz running OS 9. This is strictly a hobby. I was wondering where to go from here.

— *Felix Almentero*

Felix, Pro Tools Free absolutely provides the best bang for the buck (meaning no bucks at all), given your current computer and budget considerations. You will be limited to your computer's 16-bit audio capabilities, but this may not be a problem, since you're strictly a hobbyist. When you can afford it, consider purchasing Digidesign's Mbox. You'll get 24-bit audio and excellent mic preamps, as well as the much more powerful Pro Tools LE software, which—best of all—you'll already know how to operate.

— *Rusty Cutchin*

OUTTAKES   **Hall & Larsen**

Just wait 'til they hear *these* Oval Office recordings!

This agent doesn't hide his tracks.

Groove Agent is a stunning new VSTi that gives you ready-to-go drum tracks in just a few mouse clicks. This virtual drummer can play the most popular and influential styles from the past 50 years of music to today – all inside your VST host application. Played by top studio drummers and recorded to analog tape, Groove Agent comes equipped with high quality drum kits, both acoustic and electric, as well as a variety of percussion sounds.

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Let It Bleed

Q *I was recently reading the liner notes to my 25th-anniversary edition of Lynyrd Skynyrd's* **One More From the Road,** *and they talk about Allen Collins overdubbing his solo on the original release. I was curious how they could have avoided bleed from his live solo showing up on other tracks (the drum overheads, for instance). I was also reading the interview with the producer of Norah Jones in the JUN/03 issue, and he talks about isolation between the piano and her vocals while recording her live. How is that achieved?*

—August Rutkowski

A There are many different techniques that engineers employ to minimize bleed in live recording. First off, mics with cardioid or hypercardioid polar patterns are almost always used instead of omnis or figure-8s. These mics reject signal occurring off-axis (hypercardioids

do an even better job than cardioids), so by carefully placing and positioning instrument amplifiers and the mics themselves so that they face the desired sound source, you can drastically reduce the amount of bleed. Acoustic shields, or gobos, are also sometimes used, even onstage; a number of Plexiglas see-through models are available that are visually unobtrusive. If you're on a budget, you can make these yourselves simply by nailing together a wood frame and filling the center with a couple of inches of dense Fiberglas; you can even just drape a couple of thick blankets over the frame (packing blankets work well).

Eliminating or limiting bleed between a live vocal and piano is accomplished in much the same way. Either record with the piano lid down, or if you need to have it up, position the mics as required and then create a "tunnel" by draping packing blankets around the instrument to minimize the sound that escapes into the room. Because a singer who is playing piano is directly facing the strings, the use of a cardioid or hypercardioid is quite effective; almost all the sound from the piano comes from the 180-degree off-axis position, so little if any of it will be picked up by the vocal mic.

—Howard Massey

Back to Basics

Q *I'm a 16-year-old musician from Oshawa, Ontario. I've been fooling around with* **Cool Edit,** *and I'm interested in doing some home recording, but there's something I don't quite understand. I have a cheap little Radio Shack mixing board, and the outputs are those red and white cables like the ones you plug in to the front of a TV. Can I make it so that each channel on the mixing board corresponds to a channel on* **Cool Edit** *or* **Acid Pro 3.0?** *Where do I even plug this stuff in?*

—Seann Battams

A Seann, unless you're working only in stereo (no more than two tracks), you can't make the channels in Cool Edit or Acid show up on the channels of your mixer (you need more expensive systems with outboard hardware to have that kind of capability). But you can make some really cool mixes right in Cool Edit, mix them to stereo, and use two channels of the mixer to monitor the output from your computer. Use the mixer's other channels for instruments, CD players, or other sound sources that you want to record into Cool Edit. Of course, if the mixer has only a stereo output, you need that to drive your speakers, but you could use headphones when you want to use the stereo outputs to feed those sources into the computer. If you're using a good pair of powered computer speakers attached directly to your CPU, you can use the mixer for sound sources only. Whatever your monitoring scheme, don't scrimp on speakers. A classic rule of thumb for small stereo systems is that monitors should cost half the price of the total system (not counting software). Also, when you can afford it, try to get a copy of Cool Edit Pro, which offers much better onscreen mixing tools and other features.

—Rusty Cutchin

CAUGHT IN A TRAP

Realtraps debuts **MiniTraps** ($159.99), new membrane bass traps that also absorb midrange and high frequencies. Performance is nearly three times that of foam corners at 100 Hz, and their small size makes them easy to handle. Made with rigid fiberglass and metal instead of foam, MiniTraps are fire-resistant and can be installed in public venues. They install with one nail or hook like a picture, without glue or wall damage. *realtraps.com*
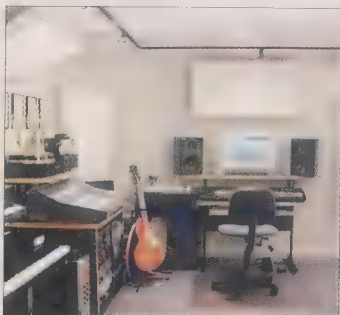
002 BE OR NOT 002 BE

Digidesign has announced the **Digi 002 Rack** ($1,295). A rackmount FireWire-based Pro Tools interface designed for recording and performance, the Digi 002 Rack features 18 channels of audio I/O—eight analog (with four mic preamps), eight ADAT optical, and two S/PDIF; integrated MIDI I/O (16 in/32 out); 24-bit/96 kHz converters; 32-track Pro Tools LE software; and DigiRack plug-ins. A single FireWire cable conveys all digital audio, MIDI data, and Pro Tools data between the Digi 002 Rack and a PC or Mac. *digidesign.com*

PHONE HOME

PreSonus is now shipping the **HP4** ($129.95), a compact headphone distribution solution featuring four discrete headphone outputs on the front panel, its own volume potentiometer, and 250 mW of amplification per channel. The high-output, low-noise (−98 dB noise floor) unit accepts balanced/unbalanced inputs (stereo or mono) and outputs (1/4" TRS), and has a mono selector button on the front channel that copies both inputs to both sides. The HP4 provides an extra pair of TRS outputs that can be used to link multiple units and feed control room monitors. Users can control the volume or mute the monitor speakers from the HP4's front panel, using up to four sets of headphones. The HP4 has a wall-mountable power supply. *presonus.com*

TRIAL BY FIRE(WIRE)

MOTU has introduced the **828mkII** ($795), a single rackspace FireWire audio interface for Mac and PC. The 828mkII replaces the 828 at the same price and adds new features, including 20 simultaneous inputs and 22 outputs, 24-bit/96 kHz operation, 8-bus CueMix DSP monitor mixing with front-panel LCD programming, stand-alone operation, front-panel mic/guitar/instrument inputs with preamps and sends, SMPTE time code sync, and MIDI I/O. MIDI timing is sample-accurate with supporting software. Additional features include ADAT sync input, word clock I/O, punch in/out 1/4" input, two 1394 FireWire connectors, and an external switch for either 115V or 230V operation. The 828mkII is compatible with virtually all audio software on Mac OS 9, Mac OS X, and Windows ME/2000/XP. *motu.com*

X MARKS THE SPOT

From **Clavia** comes a version of the Nord Lead 2 with upgraded hardware, the **Nord Lead 2X** ($1,495). Improvements include 20-voice polyphony, a large internal memory that holds four banks with 99 user programs in each, 100 user Performances, and 40 user Percussion Kits, plus 594 factory programs and 300 factory performances. Sounds can be transferred between the Nord Lead 2 and the 2X via MIDI. The unit features new 24-bit DACs running at 96 kHz. *clavia.se*

Steinberg is shipping **Nuendo 2.0** ($1,499), featuring an advanced multichannel architecture through the entire signal path. Every input, audio track, effect return, group, and output now offers up to 12 discrete channels, ready for 5.1, 7.1, or 10.2 surround productions. Several input and output busses can be used simultaneously, with any configuration possible, and any track can be routed to and from any busses. Nuendo 2.0 offers a 32-bit floating point mixer with plug-in delay compensation, real effect returns, and phase reverse and input gain change per channel. All major file and exchange formats are supported. *steinberg.net*
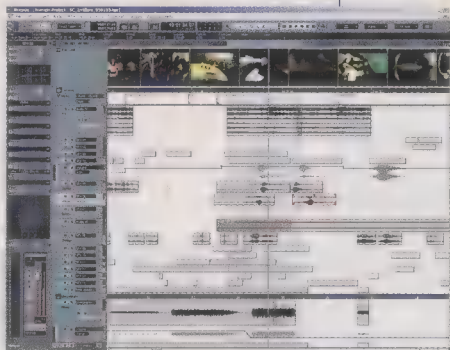
IK Multimedia has released **AmpliTube Live** ($129), a standalone virtual guitar amp and effects modeling program for Mac OS X. Providing the ability to combine three different preamp models and related EQs with three different cabinet models, it includes spring reverb and three stomp-like effects: wah-wah, delay, and overdrive. AmpliTube Live also sports a chromatic tuner. IK Multimedia has also announced that **SampleTank 1.1** (upgrade, $79) is now available for Mac OS X as an RTAS and VST plug-in for use with Pro Tools 6 and Cubase SX. SampleTank is a sampler/synth engine with 32-bit floating point processing that includes 450 multisampled sound banks (more than 2.5 GB) and supports the Akai S1000/S3000 sound format. *sampletank.com*

Roland has announced OS X support for the **M-1000** 10-channel digital line mixer. Now OS X users can fully exploit the USB audio functions of the M-1000, mixing audio from their computer along with other inputs. Up to four stereo S/PDIF inputs can be mixed—along with a stereo analog input—using simple rotary controls. The resulting signal can then be output digitally or via the analog master and monitor outputs. The M-1000 handles sample rate conversion automatically. *rolandus.com*

Native Instruments has announced that **Reaktor Session** ($249) and **Reaktor 4** ($499) are now available. Reaktor Session gives unlimited access to Reaktor instruments, samplers, effects, and live tools. It includes synthesizers and re-synthesizers, samplers and beat loopers, groove and drum machines, and surround and innovative effects. Reaktor Session also includes access to Native Instruments' online library of instruments. Reaktor 4 comes with new modules and enhanced functionality, and includes the anti-aliasing oscillators from the PRO-53. *native-instruments.com*

TC Works has released **Spark XL 2.8** ($599) with AudioUnit (Mac OS X) support. Features include a new Loop window, providing crossfade looping, loop edit tools, a set-tempo feature, and key mapping information for samplers; new file management and improved database handling in the browser view; and assignable key commands and support for long file names. The batch converter can process complex libraries with nested folders. *tcelectronic.com*

BIAS has begun shipping both the plug-in update and stand-alone edition of its noise-reduction program **SoundSoap** ($99). SoundSoap removes unwanted hiss, room noise, rumble, electrical hum, and other background noise from almost any digital media file—including digital video soundtracks, PowerPoint, Flash, DAW tracks, tape recordings that have been transferred to a computer, and other sources. The stand-alone version works with most video editing applications by opening clips, cleaning them up, and rendering a new processed file (QuickTime, .aiff, .dv, .wav, and .sd2 files are supported). *bias-inc.com*

PSP Audioware now offers **VintageMeter** as an RTAS plug-in. VintageMeter is a VU and PPM metering tool with overload indicators. All parameters, such as integration times and reference levels, are adjustable. Front panel features include displays and a VU-PPM switch. Most settings are stored when the plug-in is closed and recovered when a new plug-in instance is opened. Also new is the **VintageWarmer** ($149), a digital simulation of an analog-style compressor/limiter that comes with a library of presets. The plug-in processor can be used for both single- and multiband compression as well as brickwall limiting. It incorporates professional VU and PPM metering with overload indicators. *pspaudioware.com*

Eventide is shipping the **Clockworks Legacy** plug-ins for Digidesign Pro Tools TDM systems. The Omnipressor compressor, Instant Flanger, and Instant Phaser are modeled after analog units introduced by Eventide in the '70s. The H910 is a digital version of Eventide's first Harmonizer, while the H949 was the first "deglitched" pitch changer. The plug-ins offer full parameter automation, MIDI control, and control surface integration to take advantage of the complete Pro Tools feature set. *eventide.com*
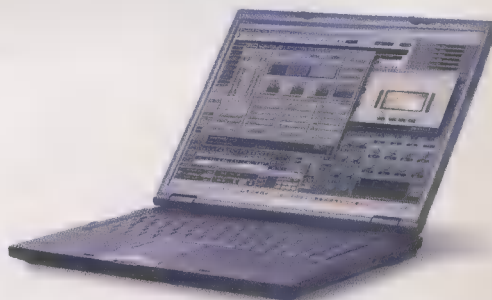
THE LEARNING CURVE

Hal Leonard has released *Loops and Grooves—The Musician's Guide to Groove Machines and Loop Sequencers* ($19.95), by Todd Souvignier. An overview of the technology and art of making music with loops, the book surveys the fundamentals and evolution of loop-based composition and explores in depth the power of current groove-making technology. *Loops and Grooves* includes a special focus on loop-sequencing hardware, a complete section on loop slicing software, and a value-added CD including audio examples and software demos. *halleonard.com*
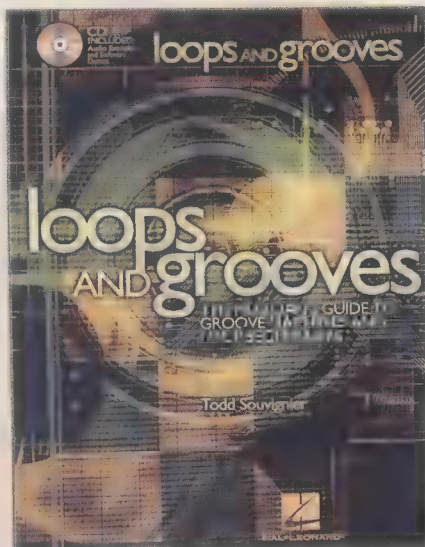
Wizoo has published *Logic: Audio Workshop* ($40), which teaches Logic users about the Track Automation system, inserts and effects, mixing, bouncing, virtual instrument and effects plug-ins, sample editing, EQ'ing, synchronization, and more. Designed for both beginners and advanced users, it includes information about computer and audio hardware, plus a collection of Internet links. The enclosed CD-ROM includes plug-ins, application demo versions, utilities, and support material. *wizoo.com*

CreamWare has introduced special educational pricing for qualified schools, music teachers, and students. The products are full versions and are available at 25 percent below list price. You must verify your academic status with the CreamWare dealer. *creamware.com*

BerkleeMusic.com started offering four online music courses in June: Pro Tools 101—Berklee/Digidesign Certification, The Music of John Lennon—The Beatles Years, Songwriting Workshop, and MIDI Sequencing Basics. *berkleemusic.com/school*

PCaudiolabs.com has announced the release of *Cubase SX Video Manual and Tutorial CD-ROM Set* ($69) for the Macintosh. The three CD-ROMs present specific topics as video clips. Disc 1 features Cubase SX menu options. Disc 2 focuses on automation, the Project Pool, the Sample Editor, MIDI and audio recording, file organization, saving effects presets, and the Devices Menu. Disc 3 gives an overview of the Key Editor, Drum Editor, Track Mixer, and Project Window; working with Hitpoints; and quantization setup. *pcaudiolabs.com*

Backbeat Books has introduced *Software Synthesizers* ($29.95) by Jim Aikin, an introduction to the world of virtual musical instruments. It covers stand-alones, plug-ins, physical modeling, FM synthesis, filters, envelope generators, MIDI, sampled loop libraries, and troubleshooting. *backbeatbooks.com*

SITES & SOUNDS

IK Multimedia and British sound designer **AMG** have launched a new SampleTank LE sound library: *Loop Soup* ($119). This double-CD set contains the loops from "Skip to My Loops" by Norman Cook, "Remix!" by Megabass, "Kleptomania!" by Coldcut, and "Pascal Gabriel's Dance Samples," plus all the loops from Now 1 and Now 2 by other AMG producers. All loops are tempo-grouped for easy sampling. *ikmultimedia.com*

Big Fish Audio has released a bunch of beat CDs. *Phat Beats From the Box* ($99.95) features New York City's beat box talent. The loops are full mixes, laid out with all the elements and breakdowns. *Methods of Mayhem 2—Damage Control* ($99.95) sports grinding guitar riffs, spooky soundscapes, darkish drum loops, and background noises. On *Scott Rockenfield's Queensrÿche Drums* ($99.95), Rockenfield re-creates the drum tracks for many of his favorite Queensrÿche songs. Each performance has been split into different loops, variations, fills, and individual hits. *HitZone* ($99.95) is a Top 40/R&B/hip-hop collection on four CDs. Discs 1 and 2 contain audio elements, while discs 3 and 4 are Acidized beat elements, ready for use in your sampler or software. *Platinum Essentials* ($49.95) has Keith Clark, aka Clizark, opening his multi-platinum bag of tricks. Heard on Snoop Dogg's "Da Game Is to Be Sold Not to Be Told," "Suited N Booted," and "Paid tha Cost to Be da Bo$$" as well as hits from B-Legit, Mack 10, Kurupt, Eastsidaz, and TRU, Clark has filled this collection with construction kits, beats, sounds, and more. *bigfishaudio.com*

From **Wizoosounds.com** comes *Claudius Bruese String Section* ($129), now available exclusively in Halion, EXS24, and Giga formats. The library includes string sections recorded at 24-bit with a natural room ambience. Featured string articulations include legato, pizzicato, spiccato, tremolo, and whole and semitone trills. *wizoosounds.com*

New from **PrimeSounds** is *Dirt Keeps the Funk* ($99.95). Inspired by the music of Parliament, Funkadelic, and other P-funk pioneers, the disc is packed with Bootsy Collins–style bass guitar lines, laid-back drumbeats, wah-wah guitars, and psychedelic synth riffs. The formats on the CD include .rex, .wav, and Audio. *primesounds.com*

David
Frank

THE MAKING OF A HIT RECORD

BY HOWARD MASSEY

IF YOU'RE OLD ENOUGH TO REMEMBER THE '80s, YOU SURELY REMEMBER THE

techno-pioneering group the System. Its innovative hits — records like "It's Forever," "You Are in

My System," and "Don't Disturb This Groove" — dominated the charts at time were in otherwise

musically bleak time and propelled the duo of Mic Murphy and David Frank to stardom. By the late

'80s, they were a major force in the industry, working with artists like Phil Collins ("Sussudio"),

Chaka Khan, Robert Palmer, and Angela Bofill and contributing to the soundtracks of two Eddie

Murphy blockbusters: *Beverly Hills Cop* and *Coming to America*.

To this day, David Frank continues to write songs and craft records that help shape popular taste.

Cowriter and co-producer of Christina Aguilera's "Genie in a Bottle," Frank is one of the most in-

demand songwriters in America. We visited with him recently at his home studio on the outskirts of

Los Angeles, where he described his unique compositional and recording process and shared his views

of how one goes about creating a hit record.

Do you generally write songs to order?

Not really. My collaborators and I just write different songs and then try to fit them to whatever artists need songs. Often we end up co-producing the songs when they're placed.

When you say you co-produce, does that mean you're doing all the playing and arrangements yourself and presenting the artist with finished backing tracks? Or is it producing in a more traditional sense?

Often I'll do a whole musical track. I did the whole thing on "Genie in a Bottle." I had already written all the music that you hear, and then we wrote the song on top of that. Many times there are a lot of things added to the basic track, but on "Genie in a Bottle," there actually wasn't—what you hear is just exactly what was there.

My impression of a lot of other production houses is that they have a lot of guys programming drums or thinking of parts, and they're all working sort of like a factory. I'm basically here just by myself. I have an assistant who sometimes helps me out with technical things, but I'm generally better off working on my own or with other songwriters; I find that my imagination is more fertile when I'm not around other people so much. That's in marked contrast to some of the other camps. They call them "camps," you know—the Rodney [Jerkins] Camp, the Max [Martin] Camp. I don't have any camp. It's just me.

So when you're co-producing, your roles are to write the songs and to create the music. What's the role of your co-producers?

Their role is doing the vocals, and they very often have a large influence over the music too, though in many cases the actual track is done beforehand; there's not much added to it, if anything. Generally, I'm playing everything on every track, but my co-producers have a lot of musical input. We'll all be present when the artist is recording the vocals and decide whether it's the right performance, and we'll comp the vocals together.

Sometimes we even record the vocals here in my home studio, though we often go to an outside studio for that. Christina's vocal on "Genie in a Bottle" was done right here. Steve Kipner, whom I work with very often, has his own studio, and our audio equipment is identical, though I have many more keyboards. When we're working together, we'll sort of bounce back and forth between studios because we get sick of being at our own houses; we just want to go somewhere else. I pretty much keep the music as MIDI data until the day before I mix so that I can keep changing it.

In the end, we record everything to Pro Tools and take the disks to an outside studio for mixing. With "Genie in a Bottle," half of that was mixed directly from Logic and half from analog tape. Until fairly recently, I had a 24-track machine here, and it sounded great—when I listen back to some of the things I had recorded on it, it has a markedly different sound. But Pro Tools sounds great, too.

What's your mixdown format of choice?

I usually still end up using 1/2" analog tape, but that's generally speaking. With [Dream's] "This Is Me," we used the 24-bit digital mix for mastering. Analog just didn't work on that particular song for some reason; the extra compression that it added was sort of overbearing. But I still like the way analog tape sounds. Part of it, of course, is what your ears are used to, and I think my ears are getting more and more used to the digital sound.

What's your vocal mic of choice?

I have a [Neumann] 147 here in the studio, and that's pretty much what we've been using, although Steve has a U 67. We'll basically try to see whether the 147 or the 67 sounds better. We've used other mics, of course. There was a custom-made tube mic that we used with Christina Aguilera, a modification that was done by Mojave Audio. It didn't sound great on everyone, but for her, it was really excellent, because it had a lot of highs and lows, and she has a very boisterous peak in her midrange. When that was reduced by the mic, it made her sound just great.

Have you done any acoustic treatment here in your home studio?

Not really, though it's something that I'd like to do. There's some foam on the wall, but that's about it. I pretty much rely on close-range monitoring, and I do a lot of my work at a very low volume, even when I'm working on grooves and things. Sometimes I'll turn it up to kind of hype myself at the end to make sure it really sounds great. Of course, I don't actually mix here. I generally mix at L.A. studios like Pacifique, Larrabee, and Enterprise, often with Dave Way.

With all the synthesizers you have here, how do you make your sonic choices when you're putting a track together?

You know, just the other night I was kind of looking around at my keyboards, and I was thinking, "Wow, this is like a little mountain of different textures." Someone might come in here and think, "Boy, he has a lot of keyboards." But to me, it's not really that many. I look at them as textural opportunities. When I'm building a track, I generally know what I might be looking for, at least in terms of a similarity to an acoustic instrument. But I also know the textural differences between the instruments I own.

For instance, the Virus is kind of a progressive texture. There's something about the sound of the Virus—it has a certain grainy quality. It's distinctly digital, but they've added some kind of warmth to it. The Nord is another useful instrument; in a way, I think of it as being somewhat neutral, though it has a vintage synthesizer texture. But it's certainly neutral in comparison to the Virus. They both have a different feeling, and they take up a different space within a mix of sounds.

The [Yamaha] EX5 is another great instrument. It's underrated, in my opinion. A lot of people don't use it, but I think it's fantastic. To my ears, it has high fidelity; there's a lot of quality to its output signal, a lot of depth. It's almost like an expensive mic preamp.

The [Yamaha] S80 is not like that. The sound quality that comes out of it is not as good, in my opinion, although I think it is a very useful instrument. But the [Yamaha] CS6R, which uses the same tone generator as the S80, is really great; it has a lot of potential for dance-oriented music. It's got really interesting ways of generating little arpeggiated patterns and things like that, yet it has a lot of edge to the sound. So I use that when I want to add a real edgy texture.

I also have two Korg Tritons—a rack and a keyboard. I have both of them because I find the factory programs to be very useful already. When I use synthesizers that have knobs on them—the Nord, things like that—I tweak them constantly, but with synthesizers like the EX5 and the Triton that don't offer so many real-time controls, I appreciate the sound design that goes into these instruments; I see that as being their strongest suit. I might tweak them a little, but I don't start from a raw sound perspective; I tend to go with the presets. The Tritons can serve as almost the foundation of what a song can be, whether it's the drums or a pad or whatever. I have two of them because, a lot of times, the programs and the effects that they put in the programs are really well thought-out. Every time you put it in combination mode and try to use it as a multitimbral instrument, you're sacrificing a lot of the things that they worked so hard to do with the effects and with the overall programming.

Years ago, I used to think, "Why would I want to use someone else's patch ideas?" But now I just realize that these are great tools for doing

what I have to do. It's sort of like giving someone else a little bit of the responsibility—and the credit—for having the music sound great. To all the guys who spend their time sound designing and making the patches in these keyboards, thank you!

Is the Minimoog your primary bass instrument?
The Minimoog was for many years my bass instrument. I remember doing the "Sussudio" session with Phil Collins in England. Hugh Padgham was the engineer, and he was saying, "I love this sound—it has a real oomph, but we can get even more fundamental by double-tracking it." But I don't use the Minimoog as much anymore.

What has taken its place?
Very often, Minimoog patches from samplers! [laughs]

Why would a sample of a Minimoog be preferable to the real thing?
Well, a lot of times, it was because the Minimoog wasn't plugged in! [laughs] If I'm in writing mode, I tend to go for the fastest sound available. I could have definitely gotten the bass in "Genie in a Bottle" on a Minimoog, but I didn't—I used the Minimoog sample out of the Triton rack. Although if you use the Minimoog itself, it definitely has more body to it. I am going to hook it up—I promise!

Do you do a lot of real-time control as you play, using things like pedals and joysticks?
Yeah, I do, quite a bit. Usually, as I record the parts, I'm turning the knobs and recording the MIDI controller data, too. It's a performance, and these are performance-oriented instruments. In fact, one of the reasons why I don't have an EX5 module is because it doesn't provide all the modulation controls. The guys who design the sounds assign specific controller information to every sound, with different setups for each. So I really need to have the actual keyboard instrument itself so that I can use what they were thinking would be great variations on their sounds.

In terms of outboard gear and audio processing, though, you seem to have a pretty minimal setup here. Does that mean you use a lot of plug-ins?
I use a lot of plug-ins [leans over to microphone and speaks in conspiratorial low tone], and I buy them all, by the way!

Have you come up with any unusual applications of plug-ins?
Well, on the Dream song "He Loves You Not," I made a lot of use of Bruno and Reso. There's a section where it sort of sounds a little like a vocoder, but there's also kind of like a high-frequency thing. I would take the odd vocal part and run it through Bruno and Reso, record it on a different track, and then cut pieces of it that I liked.

There's another song where I wanted to change the chordal structure of a vocal part, so I would record two passes with Bruno spelling out different chords, because you can latch keys on it to make the vocal be a specific chord. Then, in Logic Audio, I cut pieces out that I wanted to be one chord or another and turned one or the other track off so that it would change chords. I was sort of using them as compositional opportunities—which is what they are. There are so many possibilities of what you can do with plug-ins.

That reminds me of the auto-panning technique that some people use, where they duplicate a track, pan the two tracks hard left and hard right, and then just cut out bits wherever they want it to shift position.
Absolutely, and I do that very often. A lot of the echoes are created by

just copying a piece of audio to a separate time-shifted track and then reducing the volume. Or I might set up three or four different tracks with the same piece of audio but apply different plug-in effects on each. It can be something as simple as setting up the same echo but using a different plug-in for ■ slightly different tonal quality. On "He Loves You Not," during the guitar solo, I have a real-time echo that sounds like "*eh eh eh eh*"—not just a regular echo, but one that has a whole rhythm to it. It just came from experimenting, by moving the audio in different places on the Arrange window at different volumes and with different effects.

As someone who's been making records since the '80s, do you find that all this fine tweaking sometimes drives you up a wall? Twenty years ago, you could make a great record without having to edit at the microscopic level.
I know exactly what you mean, and I have to say that there have been times when I've been frustrated by that—because it seems like as time goes on, that sort of microsurgery gets deeper and deeper and deeper. I've had to readjust my ears to recognize it.

When I started to work with my band, the System, I had an Oberheim DSX, DMX, and OB-8 and the Minimoog. I had all the gear plugged in to a Peavey PA speaker that had a piezo tweeter in it and some kind of very cheap 15" speaker, and I would just listen in mono. I didn't know anything about high frequency or bass. I remember an audio engineer coming in and saying, "David, what are you listening to in here? Let me EQ it a little bit for you." My response was, "Why would I want that much high end? No. Shut that off; I just want it flat."

What I'm trying to say is that over the years, people have sort of turned my head and called my attention to the fact that I may have been missing some of the possibilities of subtleties. And a number of times, I've had to reorient or reeducate myself to go along with some of those things that people were recognizing. Just learning Pro Tools—it took me a long time before I actually dove into that, and I had to completely relearn everything. Before that, I had been doing everything on an MPC60 and with analog tapes. So, yes, there are those aspects. But then, of course, there are the musical aspects of having all the possibilities.

But do you think there are perhaps too many possibilities available today?
Let me put it this way: I think that in light of the amount of possibilities that there are, there's not enough really innovative music being made. You can have too many keyboards. A couple of times, I've taken a whole bunch of them out of my setup and restricted myself to working with just a few different instruments. But right now, I'm very comfortable having a lot of instruments to choose from, and I'm not overwhelmed by the amount of possibilities.

Even if you have a lot of sonic textures at your fingertips, the most important thing is that you realize the basic harmonic structure or beat or the sort of essence of what you're doing in a musical composition. If it doesn't have a real essence, it's nothing; it can just become a diffused, expendable thing. But if you spend time trying to get even one part to have real substance, that's the most important thing. That is what, in the end, allows a record to transcend all the other music that's out there, and that's what will make it popular.

How do you craft ■ hit song? What are the elements that go into it?
[*Long pause*] Wow. That's a big question. But I've had the feeling, through the years, that I've gotten closer to having an understanding of how to craft a hit song. I don't completely think that I know, of course; I can't

say that I'm confident that every time I do something, I'm going to hit the bull's-eye. But I do have more of a feeling for it than I used to.

I tend to go at composition from the standpoint of what I love. Even if it's just a beat, I have to love it, or else I don't believe that it will be a hit song. If I can think of it in any way as being common or ordinary, then it almost always really is common and ordinary. But if I can listen to something and go, "Wow, I really, really like that, and that does sound original to me and different from other things than I've heard," then I know I'm on to something. So I'm always looking to come up with something that sounds original. I might write 20 tracks, and three of them might be that great, where I feel that [they are] original enough, that [they have] enough of a harmonic basis to write some kind of emotional message over the top of [them].

I've spent most of my life listening to classical music. I love classical music; I've been taking classical piano lessons now for the past couple of years. I was classically trained when I was a kid, and I went to the Berklee College of Music. In fact, I was obsessed with practicing piano 10 to 12 hours a day for much of my life! So I have a lot of musical background, and there's been a lot of music that I've heard that I've loved—it's not isolated to pop music, or jazz, or classical. It's a blend of everything, and it's all sort of a backdrop in my mind. When I try to think of something that will be a hit song, I think of it in reference to all of those things. Some people might say, "Well, that must be a pain in the ass to always have to think that way," but I believe that I'm doing some kind of artistry, and my whole sense of reference is people that I've admired—composers and music that I've believed [were] artistry. Not business—artistry.

So even now, trying to write hit songs, I'm trying to create some kind of artistic accomplishment here. But, having said that, I have learned that a hit can be almost isolated into two separate things: a rhythm backing where people will say, "Whoa, that's a different beat than I've ever heard; there's something different about that," and then a musical track where, harmonically and melodically, people will say, "Wow, there's something about those three chords going together that makes me want to cry or makes me want to move or makes me have feelings of all different varieties." I try to find all those things: in the beat, in the harmonic structure, and in the melodic quality of the backdrop of the music.

I'm not as good with the actual top line melody and the lyrics. But over the past few years, I've gotten better. Steve has really taught me a lot about lyrics, about making a lyric into something that can be really meaningful to people. I never used to concentrate on the lyrical content of [songs]; I was always so happy when they were good, but I was never actually able to help them be good. [laughs] Now, I definitely can. Although I can never be a primary lyricist, I can really help out with the lyrics and even sometimes think of great lyrical ideas.

So in the last few years, I feel like I have gotten a much better understanding of how to craft a "hit song." Again, a lot of it has to do with meeting Steve. Steve is just the quintessential songwriter. He is a songwriter; I'm a composer. My mom held up Beethoven, Mozart, Bach—[laughs] "You could be like them!" That's sort of my orientation—to be a composer without lyrics. But Steve has a total feeling for lyric and melody being together, and he's really taught me a lot about that. I have learned to look at a song both as a whole and in its individual components—every single detail of a song. Transcending the ability of the artist to perform it, you have to be able to look at a song and go, "That is a fantastic idea." It doesn't have to be complex, but it has to really, really be something awesome in the end.

Mechanically, is that the order in which you approach things? The beat, then the harmonic structure, then the lyric, and then the top melody?
No, not necessarily. I often wake up at night and have an idea of a melody and the chords in my head, and then I come in here quickly before I forget it. I put my ideas down on a cassette so that I can wake up later and go, "Whoa, that was pretty good!"

With the benefit of 20/20 hindsight, what advice would you pass along to the reader who wants to follow your career path?
I would have to say that I really value my musical education and background. When it comes down to it, the foundation of what I do is my musical background—being able to play an instrument with my fingers. So I guess the key to my success has been listening to a lot of music and analyzing it: taking it apart, learning how to play the parts that you love, learning how to imitate all the parts on records that you think are great. Even if they are just electronic things that are derived from within the computer and they're not played, it's important to learn how to do them.

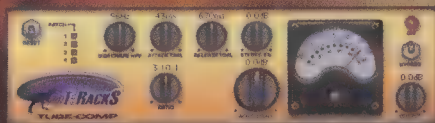*Howard Massey's latest book,* Behind the Glass, *is a collection of interviews with record producers.*

MY ENTRÉE INTO THE WORLD OF RECORD-ing in large part consisted of learning what I call "defensive engineering." For years I would do sessions where I would attempt to get a guitar sound that made me happy and comfortable enough to play my best. As I was working on the lower end of the session food chain, the engineers were often less than stellar, turning my best efforts into sounds that were often too bright and thin, and generally sucked. Slowly I picked up enough knowledge to be able to confidently say, "If you put that mic directly up against the center of that speaker cone, I will break your arm"—but with a smile, of course. An engineer friend of mine once said about guitar sounds, "If the recording engineer doesn't hear it, you ain't gonna get it." This led me to believe that I'd best learn how to record guitar myself.

In fairness to the legion of overworked, underpaid engineers I encountered, let me point out that getting a guitar to sound great on a track is not easy, especially on a quick, low-budget session. Remember, this was before amp emulation products from Line 6, Johnson, DigiTech, and the like. Even in this halcyon era of inexpensive, quality equipment and unlimited time in home studios, capturing the awesome power and/or delicate shadings of your favorite guitar tone can be a trial. To this end, the editors of *Home Recording* have rounded up 50 tips and tricks to make your guitar sound as God-like on playback as it does in the recording room. These tips can also come in handy should you be called to work on other people's projects. Just remember that if it comes down to threatening the engineer, smile when you say it. ▶
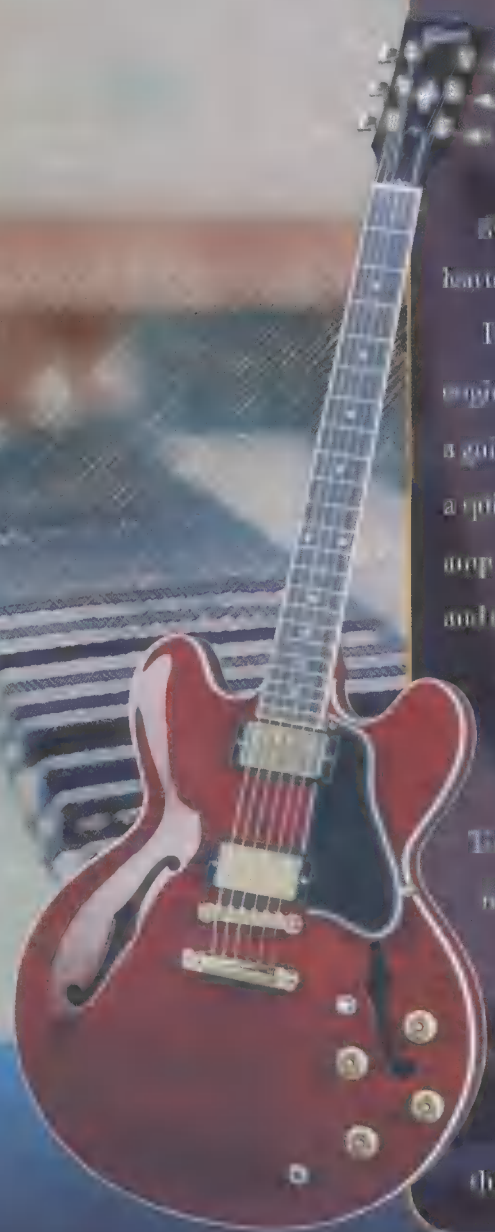
Why Doesn't My Guitar Sound Like That?

50 Tips and Tricks for Transferring the Guitar Sound in Your Head to the Tracks on Your Recorder
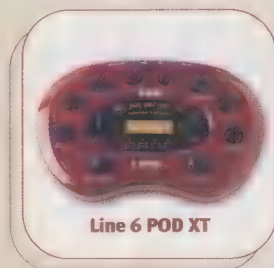
By

Michael Ross

Is It Real or Is It Modeling?

Guitarists used to dream of the day when a major-label contract with a huge recording budget would afford them the luxury of having an arsenal of guitars and amps carted to a studio. There, they would have all the time they needed to mic up various combinations in search of the perfect sound. Then they woke up.

Today's guitarist doesn't have to dream. For the price of one good guitar and amp, you can have a wealth of classic and modern tones readily available, with no cartage or miking necessary.

An assortment of amp tones can be had at the flick of a switch from a plethora of tabletop and rackmount units. The list includes Line 6's POD 2.0 Amp Modeler, POD Pro Rackmount Amp Modeler, and POD XT Amp Modeler with Effects; Tech21's SansAmp Classic, GT2 Tube Amp Emulator, Tri-A.C., and PSA-1 Rackmount Amp Emulator; the Johnson J-Station Amp Modeling System; BOSS' GP-20 Amp Factory; Roland's VG-8; Roger Linn Design's AdrenaLinn Multi-Effects Processor; DigiTech's RP 50, RP 100, RP 200, RP 300, RP 2000, GNX 1, GNX 2, GNX 3, and Genesis 3 GeNetX Guitar Desktop Processor; Zoom's 505 II, 606 Guitar Multi-Effects Processor, and GFX-4 Multi-Effects Pedalboard; the Behringer V-Amp; Korg's Pandora PX4; and, new from Vox, the Valvetronix ToneLab and Tonelab SE (Stage Edition). These units often also model or emulate different cabinet configurations as well as microphones and miking techniques (close, distant, etc.).

Line 6 POD XT

Boss GP-20

DigiTech RP 50

Move it! As mentioned in the introduction, barring extreme circumstances, you should avoid close-miking the center of the speaker cone. You don't listen to it like that, so why record it like that? Most speakers disperse low end off the edges of the speaker and high end in the middle, so move the mic until it sounds right to you.

Extreme circumstances. Maybe you actually want a very bright tone for one track. In that case, by all means center the mic right up against the cone. Or stick the amp at the end of the hall and put the mic in the toilet in a Tupperware container if you want—anything goes in the search for that elusive cool tone.

High five for lo-fi. Pristine reproduction is fine, but sometimes when you are trying to fit that fourth guitar part into the mix, bad sound is good. Cheap guitars through cheap effects into cheap amps can yield interesting textures that sit in their own space and add atmosphere (see Daniel Lanois).

Electric/acoustic. Buddy Holly's producer used to mic the strings of Holly's Strat as well as record it through the amp. This results in a track with an acoustic-style attack that can be blended in and is perfectly in sync with the electric track.

**Turn down.** If you are wondering how they get that clean, hi-fi sound for jazz or country, try turning the amplifier down to a low level (2–4). Low volume produces a much wider, more even frequency spectrum as opposed to the accentuated mids of a cranked combo.

The right tool. With effects and EQ, you can make any guitar and amp combination sound like almost anything, but there ain't nothin' like the real thing, baby. A Les Paul through a Marshall is never really going to sound like a Strat through a Twin.

Spread 'em. Delay is your friend—a 10–15 ms delayed version of the part, placed opposite to the original in the mix, will fill out the sound without wasting a track. Try a different EQ on the delayed part. Got tracks? Record a different guitar playing the same part.

Belay the boom. That expensive acoustic that sounds so good solo might not sit in a mix with other guitars and bass due to excessive low end. Try a cheaper one with less bottom—even a travel guitar like a Martin Backpacker or Tacoma Papoose.

Nashville tuning. Use a 12-string set of strings, with the low octave and double strings removed (usually acoustic, but also electric), for strumming that will sit in the mix without muddying the low end. If you combine this with a normally tuned instrument, make sure you spread them in stereo, or it will sound like a 12-string (unless that is what you want).
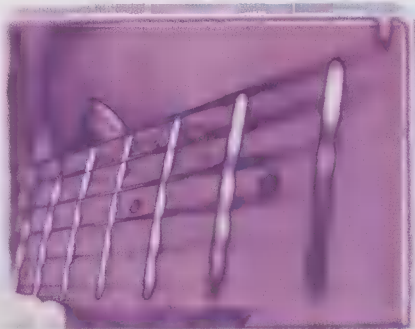
For the axeslinger who wants a variety of sounds for live performance and recording, most of these companies offer combos and heads with modeling capabilities, such as the Line 6 Vetta, Duoverb, and Flextone models; Vox's Valvetronix AD120VT Stereo 60W per channel 2x12 and AD60VT 60W Mono 1x12; the Johnson Millennium JM150, JM250, Marquis JM60, and Marquis JM120; the Zoom Fire 15 and Fire 30; and the Roland V-Series amps and Cubes. These all provide direct recording capabilities but can also be miked in the studio for real (as opposed to virtual) room sounds. At present, Fender offers no ampless device, but its Cyber Series Twin, Twin Head, and Deluxe all offer comparable amp replication capabilities and can be recorded direct.

Line 6 has recently addressed the yearning for multiple guitars with its Variax. Here, one axe offers the sound of instruments by Fender, Gibson,

Fender CyberTwin

Vox AD120VT and AD60VT Valvetronix amps

Gretsch, Rickenbacker, Epiphone, Martin, Guild, and Danelectro, as well as sitar, banjo, "Tricone," and Dobro sonorities.

Thus armed with a single instrument and a tabletop device or amp, guitarists can now turn their pipe dreams into near reality—"near" because debate will always rage as to how close these various emulations approach the "real" thing. To paraphrase the poet Marvell, had we but world and time enough to properly mic an abundance of vintage amps and go through a variety of instruments, dear guitarist, the reality would doubtless surpass the emulations. That said, the wealth of fabulous tones provided by the above products for a relative pittance, and the ease of recording them, would certainly have left the guitarists of the past aghast.

—*Michael Ross*

Line 6 Variax

Think small. A small, low-wattage amp cranked to distortion can sound as big as a stack in the studio. Parts for Derek and the Dominos were recorded through a Pignose practice amp. Two words: less leakage.

David Darlington's Tips

MOVE SOME AIR. USE A GOOD AMP to create a slammin' guitar tone before you record it. When it comes to guitar tracks, nothing sounds better than a speaker moving air. Blending a miked amp sound with your direct sound will result in a warmer and larger sound than the direct signal alone.

Tighten the space. Place the amp in a small space with few reflections (like a clothes closet) to minimize reflected sound. The direct sound from the front of the speaker cone will sound more up front in your track. Big, roomy sounds from cavernous basements with hard, reflective surfaces only tend to obscure the guitar in a track and make it hard to hear.

Get dynamic! Mic the speaker with a dynamic mic that can withstand high SPLs (sound pressure levels) without distorting. Start at the center of the cone about 6" away, and move the mic around until the sound in the control room matches the sound you hear in the booth.

Add condensation. Use a second mic, like a large-diaphragm condenser, a few feet away from the amp, and record it to a second track.

Add this track into the blend for a larger sound and a sense of space. Try panning the tracks slightly away from each other.

Hot, but not too hot. Record the guitar signals as hot as possible to utilize all the word length of your DAW/MDM or saturate your analog tape. Do not allow digital recording to go "over" the peak of 0 on your level indicator. The resulting distortion can be extremely unpleasant.

Pretreat. Invest in a good DI (direct injection) box. This box turns your guitar pickup's signal into a mic level signal for recording. Active-circuit DIs are better than passive circuits, and tube circuits are better than solid state. I recommend tube DIs by Retrospect and Avalon. The investment is well worth it.

Preheat. While investing in your guitar tone, spend money on a nice pair of mic preamps. Good mic pre's not only make everything you record sound better—not just your guitar—but really help when recording low-level signals like pickups and dynamic mics. You'll hear an immediate improvement in the "body" and "image" of your guitar tracks.

Tuning trick for computer-based recordists. Open an auxiliary track on your DAW, insert a pitch correction plug-in like AutoTune, and bus a bit of the guitar track to the aux fader. Open the AutoTune interface, and voilà! You have a digital guitar tuner that is always on and available for quick retunes anytime you're in input mode. It can also let you know if the recorded track is in tune on playback.

Keep supply lines open. Even if you're not a guitarist, keep

You can get a variety of tones and textures from one guitar by simply retuning it. Playing a part in standard tuning, then overdubbing the same harmonic structure in open G tuning (D–G–D–G–B–D, low to high), open D (D–A–D–F♯–A–D), open E (open D with each string a whole step higher), or Nashville tuning (see page 34), will thicken the sound in a way that overdubbing the identical tuning will not. If you are playing in the same octave as the original part, be aware of dissonant intervals (such as when the F♯ and G natural of a G Maj 7 chord are voiced next to each other—a half-step apart instead of 10½ or more steps apart); you may not like them.

You can lessen dissonant effects by panning the parts wider in the mix.

Restringing a guitar with heavier strings and tuning it down as little as a half-step will change its tone in an interesting fashion, adding more body. Most instruments can handle as many as two steps down. Or use light strings and go up a half- or whole step for more chime. You probably should prepare the guitar in advance, as it will likely need a new setup and an intonation adjustment. It doesn't take an arsenal of guitars for a well-layered recording—just a tuner, some strings, and some imagination.

—MR

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guitar-type supplies around for your sessions with that absent-minded player. I keep picks, a spare strap, a stand, and a tuner in my closet just in case. Also keep a pair of nail clippers and wire cutters around for hangnails and string changes.

Think stereo. Once a track is recorded, copy it, pan the two copies wide, and effect them differently. For example, flange the right side and distort the left. It will sound like the part was doubled by a great musician and will create a big stereo image on your recording.

Howard Massey's Tips

BLEND, BABY, BLEND. PLACE SEVERAL microphones up against the grille, and craft a sound from the blend of them, being careful to check phase. Famed guitarist Steve Vai—also a skilled engineer—likes to use a Sennheiser MD 421 (set to the "speech" setting so that there's no bass rolloff), a Shure 57, and a Beyer M160 for this purpose.

Spread out. Combine the sound of multiple mics in a wide stereo image. Vai often pans the Beyer either hard left or at 9:00 and pans the other two mics at 2:00.

Ace the space. Use room mics for ambience. Vai favors a large-diaphragm condenser such as an AKG 414 about 3' in front of the amp, and also blends in the sound of another couple of ambient mics placed in the corners of the room.

The delay way. Try delaying the room mics by 20 ms or so to create extra dimensionality, almost like an ADT (Automatic Double-Tracking) effect. You can also try reversing the phase of one of them—Vai often does that to the mic positioned 3' from the amp.

Double-track! Try to play as tightly as possible—the inevitable human errors will still be there, bringing the sound to life. You can create chorusing effects either by slightly detuning the second guitar or, if you're using tape, by slightly slowing down or speeding up playback during the overdub.

Get in the back. A great way to add a lot of body and depth is to mic the back of the guitar amp speaker as well. Generally, the microphone in the back has to be positioned so that it's an exact mirror image of the mic in the front; if it deviates even a little bit, it probably won't be in phase.

Compress and connect. If you use a multiple-mic technique, strap a compressor across the combined signal (via a bus or subgroup) rather than the individual mics. This will help "glue" the individual components together into one overall guitar sound.

Move out. Try placing the guitar amp in different acoustic environments—in a tiled bathroom, in the bottom of a stairwell (with an ambient mic positioned at the top), or in the center of a room (as opposed to up against a wall).

Lean on me. Tilt the speaker cabinet. Leo Fender put legs on the sides of the Twin so that the guitarist could better hear himself, but the other consequence is that when you put a mic up against an amp tilted that way, the angle of incidence changes, so you don't get phase cancellation problems off the floor and wall.

Change directions. Try recording acoustic guitar with an omni, as opposed to cardioid, mic. My personal favorite mic for this application is the Audio-Technica 3527, but you can get good results with most small-diaphragm omni condensers. Their lack of proximity effect eliminates the unwanted boxiness and boominess that many cardioid mics add when used in this application.

David Simons' Tips

BEEF IT UP. POINT A SHURE SM57 directly at the speaker cone, then add a Sennheiser MD 421 (or some similar model) off-axis for extra bottom.

Give it room. Place an SM57 against the grille (slightly off center). Then situate a large-diaphragm condenser mic several feet back from the amp. Mix in stereo and adjust pan settings to taste (and watch out for phasing!).

Pan it with echo. Move your guitar track off to one side of the mix. Then add a splash of stereo reverb with the return delayed (so that the guitar echo "jumps" to the opposite channel)

Get small. Forget the Marshall stack. For great tone (and real amp crunch to boot), go with a small, low-watt tube amplifier outfitted with a crusty old 10" Jensen.

Re-EQ it. While it's preferable to set your tone at the source, you can achieve interesting results by radically manipulating the signal at mixdown using board EQ (i.e., jacking up the mids and rolling off the lows will give you a direct, punchy sound, suitable for lead work).

Compress it. Particularly if you're pumping up the volume, run your electric-guitar signal through a compressor at a 3.1 ratio—but unless you're seeking a radical effect, keep the threshold in check.

Let it bleed. Though some isolation is necessary when recording electric, don't overdo it. Letting the sounds mingle a little will produce a much tastier mix.

Let it feed. A tube amp (or tube overdrive box), combined with a hot pickup, can produce some tasty harmonic feedback. But for the best

sustain, situate yourself at floor level near the amp so that the pickups are directly in line with the speaker (closed-cup headphones are required).

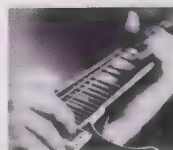**Echo from the amp.** If amp reverb is available, use it. Unless it's a real cheapo effect, it'll still sound better than mixing the signal with board reverb later on.

Trash it. To make your little amp sound like a little monster, get a large galvanized trash can, stick the amp inside, insert a microphone, then adjust the volume to taste (cover optional).

Thad Brown's Tips

USE VOLUME SWELLS TO MAKE PAD sounds. Roy Buchanan was a master at using the volume control on his guitar to create amazing effects. Once you've learned a bit about that technique, all kinds of things are possible, but a cool one is making something like a synth pad with a guitar. Swell up individual notes or a whole chord, add lots of delay with high feedback, and run the result through a swirly phase shifter.

Capo double-tracked parts. Guitars are some of the most commonly multitracked parts in music production. Usually a player will double rhythm parts and pan the multiple tracks to thicken the part. An interesting way to add to this is to get a capo and play the same chords, but in a different position on the neck.

Use alternate tunings. Lots of bands detune these days to get that massive Korn sludge guitar sound, but think about other tunings. Try tuning to a minor chord or in one of the famous blues tunings, like open D or open G. DADGAD is another great alternate tuning; you can get a great open tuning sound but still play convincing "cowboy chord" parts.

Buy a cheap 12-string electric. With their natural detuned sound, 12-string guitars are a great addition to any guitar-recording arsenal. They sound great as chimey rhythm guitars, do nicely for playing slightly overdriven hooks, and can even sound good through distorted amps if they are in tune and played well. While some are expensive, Danelectro makes a great-sounding 12-string for only a few hundred dollars.

Put away the Tube Screamer. Yes, Stevie Ray Vaughan had great tone, but that doesn't mean every guitar solo needs to be played through a distorted tweed amp being overdriven by an Ibanez Tube Screamer. It has become the blues/rock equivalent of using the Funky Drummer breakbeat in a hip-hop song—tired and used up. Try other distortion pedals, fuzz pedals, or none at all.

Try different speaker cabinets. Most people record guitars through combo amps. There's nothing wrong with that, but try different speakers and cabinets with your amp. Usually you can just run a speaker cable from the back of the amp to a different cab, letting you try out any cab.

Use different pick types and gauges. Guitar players spend huge amounts of money

THE POWER OF

LOGIC AUDIO

The PC Door Closes, but
Logic's Future Is Wide Open

BY JOE ALBANO

emagic

Logic|6

Music Production Software
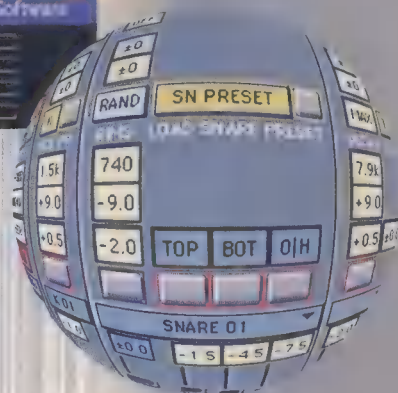
IN THE EARLY '90S, EMAGIC WAS FORMED FROM THE ASHES OF C-LAB, A German company that made an excellent sequencer for the Atari ST platform called Creator/Notator. The new company's first product was Logic for the Mac. Originally a Notator user, I was working with most of the major Mac workstations of the day by that time, but when I took a look at some of the advanced features of Logic's MIDI section, the flexibility and stability of its audio engine, and its highly customizable interface, I switched over and have been using Logic as my primary workstation ever since.

Over the next 10 years, Logic went dual-platform (Mac and PC) and grew into a "killer app," and Emagic hummed along. Then, in 2002, the industry was taken by surprise when Emagic was bought by none other than Apple Computer. Though Apple has stated its intention to ▶

maintain Emagic as a completely independent "wholly owned subsidiary," the one big change has been the discontinuation of Windows development, returning Logic to its Mac-only roots. Windows users are understandably upset, but Logic's Mac user base has settled in, hoping that the return to a single platform, along with the benefits of Apple support, will result in faster development cycles for new features and even more stability and power.

Although the features of the various music apps have begun to overlap, Logic still has unique capabilities. There are some things I regularly do in Logic that I just wouldn't be able to do in any other program without additional software or hardware. Let's take a look at what makes this powerful program such a great environment for making music.

THE BASICS

In its current configuration, Logic (like most DAW software) is offered in several versions: the top-of-the-line Logic Platinum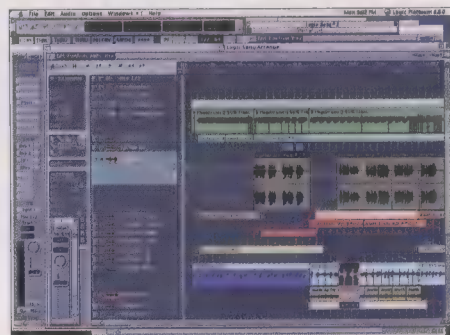 ($949); Logic Gold ($649), with a slightly reduced feature set; and Logic Audio ($299), a simpler, more entry-level option (currently available in the "Big Box" and bundled with a selection of Emagic software instruments). Emagic has a comparison chart of the different versions on its Web site (*emagic.de*). In this article, I'll be describing the features of Logic Platinum.

One of the most important aspects of any program is its user interface; this, as much as the feature set, is a main factor in a user's preference for a particular DAW. Logic has always offered an unprecedented amount of user control over layout and operation; this has been one of its great advantages, especially for users who have fixed ideas about how they want to organize their workspace. Like most current DAWs, Logic seamlessly integrates MIDI sequencing and audio recording. MIDI and audio tracks appear together in the main window, called the Arrange window (see **Fig. 1**), made up of the list of tracks and a time line, where recording and song editing are carried out. When you're ready to mix, you can use a separate window, the Track Mixer, with its virtual console of faders and knobs.

Fig. 1. A typical Logic Screenset, containing a Transport strip, 1-channel Track Mixer, Event Float, and Arrange window with MIDI and audio Sequence Objects. Also visible are both track-based (top) and object-based (bottom) automation curves. Below the Transport controls are a string of user-created Environment controls that add functionality.

In Pro Tools (developing originally from the audio side), these two windows are it—and this simplicity appeals to many Pro Tools users. However, as with all of the integrated audio/MIDI workstations (Logic, Cubase, Digital Performer), Logic has many powerful MIDI editing features, which require additional windows: the Matrix window (a "piano roll" time line editor for graphic display of MIDI notes and controllers), the Score window (a display in standard musical notation), the Edit List (a text list of MIDI events),

LOGIC HARDWARE

As a "native" audio system, Logic depends on the computer's CPU to handle most of the processing for audio recording and playback, as well as for running plug-ins and virtual instruments. Fortunately, most computers nowadays are up to the task, allowing you to create some awfully complex tunes, especially with dual-processor models—and it's only getting better.

If the computer has audio I/O onboard, that (plus extra RAM and hard disks) could in theory be all you need. In reality, most onboard A/D/A converters are noisy and not up to par for serious audio use. An add-on interface with anywhere from 2 to 24 channels of analog and/or digital I/O is what the job calls for. Fortunately, there's no shortage of options, and the good news is that Logic supports pretty much everything. Logic is compati-

Emagic EMI 6|2m USB audio interface (top) and AMT MIDI interface

ble with all the major audio hardware driver formats; it can be used with the computer alone (Mac AV), its own Audiowerk PCI card or EMI 2|6 and 6|2m USB interfaces (EASI), or hardware from MOTU, RME, M-Audio, etc. (ASIO), as well as Digidesign (Direct I/O or TDM). On OS X, Logic will support any audio interface that has a Core-Audio driver.

All of the above are native systems except, of course, TDM. An add-on package from Emagic allows Logic to be used as a front end for full-blown Digidesign TDM and HTDM systems, while the ESB Logic-TDM bridge integrates native-processed audio plug-ins and instruments with the TDM mixer within Logic, allowing for the full use of all of Logic's internal goodies in the TDM environment—the best of both worlds.   —JA

and the Hyper Edit window (a graphic display of user-definable groups of MIDI controllers in a track). There are also separate windows for audio file management (Audio window), destructive audio processing (Sample Edit window), and transport and other controls (Transport window); a dedicated mixing window (the Track Mixer); and the Environment, a unique feature of Logic that allows you to manipulate MIDI and control signal routing (more on that later). With all these separate windows, one could easily spend as much time opening, closing, and rearranging them on the screen as getting actual work done, and I've often seen novice users on various DAWs doing just that. But Logic really shines because of a brilliant window-management feature called Screensets (**Fig. 1**).

SCREENSETS

Logic's Screensets feature has proven so popular that most other programs have instituted a version of it, but Logic's implementation is still the best. Screensets allow you to create an arrangement of windows on the screen to facilitate a particular task, and then save that arrangement, lock it, and assign it a number on the computer keypad. Each numbered Screenset can be optimized for different uses and called up at the push of a button. Going from a screen set up for tracking to one for mixing to one for MIDI editing takes a second, and there is no need to reposition and drag windows around before you can start working on a particular edit or mix move. Windows can be "linked" within a Screenset, so a note editor opened next to an Arrange window can be set to display the content of selected sequences in Arrange, automatically follow the selection of different sequences, update its display on the fly, and follow as the user jumps to different sections of the song. You can even open multiple views of the same window (e.g., several Matrix windows showing the contents of different tracks, or a couple of Arrange windows showing different views of the

song). All of these different setups can then be saved into separate Screensets, 90 in all. Once a user has set up appropriate Screensets, they can be stored in a user-designed template song (which Logic calls the Autoload) and be available to every project. New Screensets can be added and saved with individual songs or be imported from song to song. It's a tremendous workflow enhancer—my Autoload has a number of customized Screensets that really let me fly around the program during sessions.

THE ENVIRONMENT

Another unique feature that contributes to Logic's flexibility is the Environment. This is a graphic representation of the MIDI and control-signal flow within the program—the behind-the-scenes part of any DAW application that is usually fixed and invisible to the user. In Logic, the Environment can be reconfigured to allow users to modify MIDI signal flow, implement processing, and add custom controls to the workspace. Some typical examples would be creating MIDI filters to adjust velocity or controller curves on record or playback, designing custom control panels to automate internal features of external sound modules, designing musical processors (like arpeggiators) that can act on MIDI and virtual instrument tracks, adapting external control surfaces to various functions for performance or mixing (see **Fig. 2**)— the possibilities are vast. I have used my custom-designed Environment patches not only for my own studio work, but also to add functionality or simplify access to certain features for some of the novice Logic users I consult for. For users who are not comfortable messing around "under the hood," so

MY FAVORITE PLUG-INS

Logic's channel strips can now load up to 15 plug-ins, with full automation control. Besides supporting AudioUnit and VST formats, providing access to all my favorite tools—such as Auto-Tune, the Waves Renaissance Comp/EQ/Reverb and C4, and a host of others—Logic comes bundled with an impressive collection of plug-ins that can be used only within the program. The selection runs the gamut from dynamics, EQ/filters, delay-based effects, reverbs, and mastering processors (Multipressor) to sound-mangling processors (Distortion, Overdrive, BitCrusher), utilities (Gainer and DirMixer, a stereo enhancer and M/S decoder), and specialty items like the AutoFilter and Spectral Gate, the Enveloper, and the new EVOC, a full-featured vocoder. Some of my favorites are the Channel EQ (new in Logic 6), an 8-band model that rivals the best of the high-end outboard EQ plug-ins for warmth and transparency; the Tape Delay, great for getting those old-fashioned echo effects we all know and love, either free-running or synced to the song's tempo; the Enveloper, a Transient Shaper that can adjust Attack and Decay (I once used it to salvage a great bass part played on a miserable-sounding bass); and the DirMixer, nothing fancy but an invaluable tool for adjusting the width and spread of stereo tracks. From the many freeware and shareware plug-ins that also work beautifully in Logic, my fave is the free MDA (Maxim) Combo, a simple but elegant amp/speaker simulator, available in VST and AU formats, that sounds great with a Rhodes or Wurli sample pumping through it. —JA

(Counterclockwise from left) Emagic Channel EQ, Tape Delay, and DirMixer; MDA Combo

to speak, there are many preassembled Environment patches available online that can, with just a little effort, be copied and pasted into any Logic song. These are designed by Logic experts and are freely available (for the most part) to anyone who chooses to take advantage of the extensive online Logic users community (use the links on Emagic's Web site or search engines for Logic-related resources and discussion forums; see the sidebar "Logic Support" for more details).

THE ARRANGE WINDOW

Logic's main window for recording and editing is the Arrange window. As with every DAW, this is laid out as a list of tracks (audio, MIDI, virtual instruments, video) against a time line (bars/beats, SMPTE). Recorded material is displayed as horizontal bars that can be divided into sections (in Logic's terminology, Sequence Objects) for editing. To the left of the track list are the Toolbox and Parameter boxes. The lower box, the Instrument Parameter box, allows for playback and display settings for the selected track's voice (such as transposition per MIDI channel or display settings per audio channel). But Logic also offers finer control. The upper box, the Sequence Partbox, allows for independent playback settings for individual regions in the Arrange window (for example, different sections on the same track can have different nondestructive settings like quantization or velocity offset). This takes some getting used to for those coming from DAWs where all such controls affect only entire tracks or voices, but it provides a great deal of flexibility.

EDITING IN LOGIC

Editing of regions (Sequence Objects) in Logic is fast and intuitive. There is the usual array of tools: pointer/arrow (move Objects), scissors (divide/cut Objects), pencil (create Objects), eraser (delete Objects), I-beam (name Objects), mute, solo, crossfade, and some automation controls. When it comes to moving Sequence Objects around, rather than making you choose from a number of modes to determine the behavior of dragged Objects (à la Pro Tools), Logic has always taken the approach of offering one main mode (the pointer/arrow tool) along with a number of modifier-key options and key commands. For example, as befits its MIDI-sequencer origins, Logic automatically snaps dragged Objects to bar/beat divisions (in version 6, this is zoom-dependent as well); for greater resolution, holding down Control while dragging enables a finer (user-definable) grid, and adding the Shift key sets the grid to the highest resolution possible. User-definable key commands can also nudge Objects back and forth. To make Sequence Objects automatically butt up against each other with no gaps, select the Objects (still with the arrow tool) and hit the appropriate key command. To drag the left or right edge of an Object, simply position the arrow above the lower left or right corner until it turns into a finger, and then drag as above. To make quick, safe cuts, the scissors tool can be assigned to the Command key as a momentary modifier—while Command is held down, the arrow turns into a scissors. Dragging through a sequence will scrub it; releasing the mouse button will make the cut and automatically revert to the arrow tool, preventing further accidental cuts. A new tool, the Marquee, has also been added to the Arrange window. This allows for selection of regions inside one or more Sequence Objects simultaneously, which can then be further edited together (this capability is offered in other DAWs and has been welcomed by those who move back and forth between various systems).

Logic's approach to editing is consistent in all the editing windows, so dragging notes in the Matrix Editor works the same way as dragging sequences in Arrange. For even more precision, a linked Event List will allow for numerical entry as an alternative to dragging—in fact, the Event Float is an unobtrusive one-line floating event display that shows all the stats for the current selected event in any window (Sequence Object, MIDI note). It can easily be squeezed into any Screenset, providing a readily available editable display that's always on hand for fast, accurate adjustments.

To further facilitate editing work, Logic also offers a variety of zoom options (including individual track zoom), a recently introduced Undo History, and a complete set of fully user-definable key commands, which can be imported from any Logic Preference file. (For those who must bounce back and forth between different

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LOGIC SUPPORT

There is a lively community of Logic users that can be accessed through a variety of Web sites and discussion groups. Following the links on Emagic's Web site can take you to most of these. Here are a few noteworthy Logic resources:
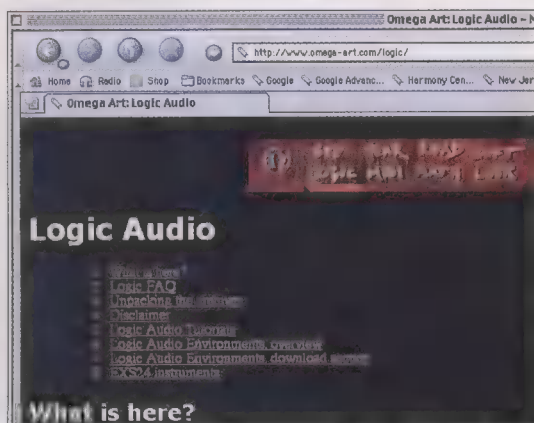
Yahoogroups.com. Yahoo sponsors quite a few Logic-related discussion groups, either on this Web site or via e-mail, including the LA-Mac list and the EXS users list. The biggest by far is the Logic users list, with more than 11,000 subscribers. This excellent source for information exchange is visited by a lot of extremely knowledgeable Logic users who are willing to share their expertise, and there's occasionally some (unofficial) participation by Emagic programmers.

Swiftkick.com. Len Sasso maintains a comprehensive (perhaps the definitive) collection of Environment patches here, as well as many other Logic-related goodies.

Omega-art.com/logic (pictured). Hendrik Jan Veenstra maintains a highly informative Logic FAQ, full of excellent tips and tricks assembled from various sources (often things you won't find in any manual), as well as Environment patches and a lot of other very useful information. *—JA*

programs and want to be able to maintain some consistency, Logic even offers key-command sets to match other DAWs.)

Logic's editing style, once you get comfortable with it, is fast and easy. There's more that makes Logic a great tool for creative tweaking, but let's jump ahead and take a look at what Logic offers in the areas of audio and mixing.

LOGIC AND AUDIO

Logic's audio engine has grown into one of the most complete systems around. It has all the usual features you'd expect: massive track count, support for all the main audio file formats (.sd2, .aiff, .wav), a full-featured mixing environment, a powerful automation system, support for VST (OS 9) and AudioUnit (OS X) plug-ins and virtual instruments, and a comprehensive set of internal plug-ins and instruments that may be the best of any DAW's built-in offerings (see the sidebars "My Favorite Plug-ins" and "Virtual Instruments").

The Logic audio engine is a "native" system—it uses the computer's CPU power to handle all the processing, rather than dedicated hardware like Pro Tools TDM packages (although Logic also supports TDM systems). Fortunately, computers nowadays have enough power to run even large audio setups reliably at low latencies, and with Logic's support of a wide variety of audio interfaces, there's a healthy range of audio system options (see the sidebar "Logic Audio: Hardware").

Within the program, managing audio files is a breeze. The Audio window provides the tools for organizing and keeping track of the various files that accumulate during sessions. Logic 6 introduces the Project Manager, a comprehensive database within Logic that provides intelligent file management and assists in creating backups and archives.

Nondestructive editing of audio in the Arrange window (handled just like MIDI data) is described above. Besides the tools and techniques already mentioned, there's a crossfade tool, which allows you to quickly add nondestructive fades between adjacent sections of audio in one swipe. The Sample Edit window offers a number of destructive DSP tools such

as normalize, gain change, trim, and sample rate conversion, as well as more specialized functions like pitch shifting and time stretching. So right from within Logic itself, you get most of the functionality of programs like Peak, eliminating the need to bounce between applications for this kind of processing.

MIXING IT UP

There's more than one way to do just about everything in Logic, and mixing is no exception. There are two different ways to view mixer channels: in an Environment window, where you can graphically arrange audio channel strips (Audio Objects) and MIDI faders and knobs into a custom-designed virtual mixer surface, or in the Track Mixer window, where the display of channels is created automatically, following the track list in the Arrange window (see **Fig. 3**). The Track Mixer is easier to use; you don't have to do anything except choose from a few basic options. You pick what type of channel strips you want to display (MIDI channels, instrument tracks, audio tracks, buses, auxes, masters, etc.) and whether you want the order to be fixed (e.g., all audio channels displayed in numerical order, as on a hardware mixer) or to dynamically follow the order of tracks in the Arrange window (so grouping together,

VIRTUAL INSTRUMENTS

Emagic has done a bang-up job of developing its own selection of Logic-only virtual instruments. The EVP88 nails various Rhodes models, Wurli, and Pianet. Its musical dynamics and touch response make it sheer fun to play. Likewise, the excellent Hammond/Leslie simulation of the EVB3 and the spot-on Clav sounds of the EVD6 are a blast. I haven't logged as much time using the ES1 and ES2 synths, but my experiences with them revealed fat-sounding, very flexible synths. My personal favorite is the EXS24mkII. With the mkII upgrade and the addition of a powerful matrix modulation section and other enhancements, this little sampler—already very popular for its straightforward interface and efficiency—has almost come into its own as a full-fledged virtual sampler. It needs only better multitimbral capability and a slicker interface for Sample Instrument editing to be complete.

(Clockwise from left) Emagic EXS24, MDA Piano, and Emagic ES-P synthesizer

The above Emagic instruments are integrated into the program but sold separately. Included for free are the ES-M, ES-E, and ES-P synths—a mono bass synth (M), a pad (ensemble) synth (E), and a polyphonic synth (P), respectively. They sound good and come with a nice set of presets to get you started. All in all, it's an excellent set of music-making tools. Add the variety of third-party instruments supported, such as Native Instruments' Kontakt and B4 and IK's SampleTank, and you've got an almost completely self-contained music machine. Just hook up a controller and wail. *—JA*

say, a number of harmony vocals on audio channels 1, 4, 6, 9, and 12 in the Arrange track list will result in those channels being in the same order side by side in the Track Mixer). The audio channel strips in the Track Mixer are the same ones as those in the Environment (created when you first set up the program)—adjustments to a fader in one window are reflected in the same fader in the other. It's a bit more work to set up an elaborate virtual mixing surface in the Environment, but you have more control over the layout. Both virtual mixer displays are useful; I often find myself moving back and forth between them during a mix. Another mixing convenience is the option of having a single channel strip (either a narrow Track Mixer or Environment window, one channel wide) linked to the Arrange window. This linked "floating" channel strip will follow the selection of tracks in the Arrange window's track list, allowing for quick access to any mix adjustments when there's no room for the display of a full mixer. This idea has proved so popular that Emagic recently incorporated a floating channel strip directly into the Arrange window, under the Partboxes (for those with big enough monitors). Either way, it works great and is a major time-saver.

Fig. 3. An Environment mixer panel of audio channel strips (left); a Track Mixer with MIDI fader strips (right)

AUTOMATION

Logic's original automation scheme allowed for MIDI mix data like Controller 7 (volume) and Controller 10 (pan) to be recorded within Sequence Objects along with the MIDI notes or audio, and displayed as a continuous curve with breakpoints (HyperDraw display) in the Arrange window, on top of the sequence data or waveform display. While this Object-Based Automation worked fairly well, the use of the 127 MIDI controllers for that purpose fell short when it came to automating a large number of controls on a large number of plug-ins and virtual instruments, so Emagic developed a Track-Based Automation scheme and a new type of internal automation data called Fader Events (**Fig. 1**). The new data type allows for full control of every parameter on every plug-in slot, and the track-based scheme is a more industry-standard approach, familiar to users of console automation systems and other DAWs. Standard automation modes are present: Read, Write, Touch, Latch, and a mode for MIDI messages. Along with the new system came improvements in responsiveness, simultaneous display of multiple mix curves for a track, many new controller-editing features like exponential- and S-curves, and tight integration with Emagic's Logic Control moving fader control surface (Logic 6 adds support for many additional control surfaces as well). The new Automation system is extremely slick—the first mix I did with it (a surround mix with lots of stuff flying around the speakers) was done in half the time I'd expected, and that was just using the mouse. Finally, for situations where the original Object-based approach still makes the most sense (like automating controls on a virtual instrument as part of a musical performance), the old system is still available, enhanced with most of the new features and improvements.

A Bounce button on the master outputs allows you to mix down internally to stereo or surround, either in real time or "offline" (faster than real time, as long as no external sound sources are feeding into the mixer). A choice of high-end POW-r dithering algorithms is available for bounces of finalized, mastered mixes. In OS X, bouncing to .mp3 is also available.

THE WRAP-UP

This brief description of Logic naturally doesn't come close to covering all the special features and advantages the program has to offer (for more details, see the review of Logic 6 on page 56). It's one of the deepest applications of its kind; there's always something you can learn or discover, even after working with it for years—some function you didn't think the program had, but then you find out it does. Emagic seems determined to keep Logic on the cutting edge; it's one of the most complete DAW implementations out there, yet new features keep being added. And despite the complexity that naturally results from giving the user so many options and choices, basic operation is fast, efficient, straightforward, and, well, logical—you don't have to jump into the deep end to do good work in Logic, but it's there when you're ready to really get your feet wet. ▌▌▌▌

Joe Albano is co-owner of Rooftop Productions (rooftopproductions.com) and a freelance producer and consultant in New York City.

GETTING REAL

Adding Appropriate Ambience to Your Recordings

IT'S VERY TEMPTING TO LOOK FOR A "MAGIC BULLET" WHEN IT COMES to crafting your sound. With a minor cash outlay, you can have access to virtually any sonic tool imaginable in a single rack or plug-in space. Compression, reverb, delay, phasing, equalization, limiting—I could name manufactured sonic tools from A to Z. There would, however, be one tool missing, and that tool is reality.

Achieving a "realistic" sound may not always be the goal, however; in fact, sometimes you may want to suspend reality or create an alternate reality, considering that certain varieties of popular music are produced entirely in the electronic domain. There is no acoustic reference for the way a bank of synthesizers and drum machines are "supposed" to sound in real space, since they never existed in real space.

Yet what separates great recordings from not-so-great recordings is a sense of space and depth. For those of us working in project or home studios, achieving "real"-sounding recordings can be a challenge. Lacking the appropriate environment, instrumentation, and players can put a definite crimp in your finished product. Still, it is possible for a home or project recordist to create tracks that give the illusion of real space, and in this article I will illustrate some strategies to help you bring new life to your projects.

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Hi-Fi, Lo-Fi

Spend an afternoon listening to your grandmother's records. If that's not an option, head to your local library and choose a random stack of records (or CDs) from the '30s to the '80s. Try to include records of all different genres, from symphonic to rock 'n' roll. As you move forward in time, you will notice that recordings tend to increase in fidelity from the '30s to the late '50s and slowly decrease in fidelity from the late '60s onward. While there are myriad reasons for this, including the transition from vacuum-tube to solid-state recording equipment as well as changes in the recording spaces themselves, the biggest factor, in my opinion, is the introduction of multitrack recording.

What? Multitrack recording has negatively impacted the fidelity of recordings, you say? Are you nuts? Relax. Multitrack recording is obviously here to stay and is an essential part of the modern music-making process. I will not be coming to your studio to confiscate your ADATs or Pro Tools systems! Just the same, consider the different ways that multitracked (and, to a lesser extent, multimiked) recordings have conspired to change the nature of recordings.

BY RICHARD ALAN SALZ

ILLUSTRATION BY JOAN HALL

The most obvious change that occurred was that no longer did all elements of a performance have to take place at the same time. Although this has given producers and musicians incredible freedom, it does depart from the tradition of ensemble playing and can result in a track lacking a sense of cohesion, both musically and sonically. Multitrack recording also gave rise to the practice of using different recording spaces (and studios) from track to track, further confusing the sonic environment.

In the early days of recording, it was common to use one or two microphones to capture an entire performance. Engineers, often through trial and error, would find the location for the musicians and the microphones in the recording studio that yielded the best overall balance of sound. Later on, as recording technology became more sophisticated, it became common to group-mic instrumental sections and use spot microphones to highlight specific instruments, especially in orchestral settings.

Multitrack recording also brought about changes in the basic way that instruments were miked. To achieve more isolation (making the process of punch-ins easier), microphones were moved closer to the instruments. Some instruments were direct-injected into the mixing console. All of this moved the recording process further away from being a documentary of a musical performance in a given time and place.

That is to say not that these changes are bad, only that they depart from previous recording and performance practices. Consider the current dominant recording paradigm on an instrument-by-instrument basis for a typical recording session:

Drums: Individual microphones on kick, snare, toms, hi-hat, cymbals, overhead mics, room mics
Electric guitar: Close-mic on amp, room mic(s)
Acoustic guitar: Close-mic with small-diaphragm condenser(s)
Bass guitar: DI track, close-mic on bass amp
Percussion: Close-mic (usually with small diaphragm)
Keyboards: DI tracks
Vocals: Close-mic with large-diaphragm condenser
Background vocals: Close-mic with large-diaphragm condenser

Top 10 Strategies for Getting Real

1. Move the microphone rather than use equalization.
2. Invest in microphones with excellent off-axis performance.
3. Use the minimum necessary number of mics/inputs.
4. Leave room mics in position for the entire project to enable you to add room sound to any multitracked instrument you use.
5. Use natural reverb and echo spaces when possible.
6. Don't be afraid to commit to an ambient-sounding instrument.
7. Arrange overdubbed instruments in natural positions within the room (as in a live performance).
8. Keep overdubs to a minimum; make basic tracks "keepers."
9. When mixing, use ambient sounds as the primary sources, introducing close microphones just to add impact.
10. Use compressors to bring ambient sound forward, gates to minimize recorded ambience.

When you compare this setup with the old miking style of using one or two microphones for an entire recording, it's not hard to see how an accurate perception of tonal and spatial reality is compromised. Still, it is often deemed essential to have control of each element in a recording. This obviously results in two diametrically opposed recording paradigms.

Typical home and project setups might have a pared-down microphone count, but they can still suffer from "over-miking" as well as a buildup in mic "sonic signature" when the same microphone is used on track after track of overdubs.

All Miked Up and No Place to Go

While the idea of recording straight to 2-track is a worthy one, for many project and home recordists, it is simply impossible to conceive of recording a project without overdubs. Often one musician will be performing multiple parts in a particular arrangement, or there may be a shortage of

space and equipment. That's okay; it's still possible to bring a sense of "air" and depth to many recordings.

Even purpose-built studios can sometimes be suboptimal when it comes to getting large- and open-sounding recordings, especially when microphones are placed in the "conventional" close-miked positions. The key to recording in any space is understanding that different acoustic spaces will respond differently to the same instrument(s). That said, here are some things that will apply to any space you may be working in.

There will always be a "vanishing point" where the amount of reflected sound begins to eclipse the amount of direct sound. This is often a great starting point for microphone placement, as minute adjustment of the microphone can radically change the sound of the instrument.

Low frequencies always build up in corners. Instruments that are somewhat shy in low frequencies can be placed in the corner to emphasize their low-frequency output or, conversely, be moved away from corners to diminish their low-frequency output. Similarly, microphones can be placed or aimed toward room corners (this includes floor-to-wall and wall-to-ceiling junctions) to obtain a boost in low frequencies.

Hard surfaces such as plaster, stone, and some kinds of wood will produce brighter sounds. If there are large quantities of these surfaces, sounds will tend to reflect off them, producing reverberation. Conversely, soft substances will tend to kill reflections and produce dry and dead sounds. Obviously, the optimal recording space includes elements of both.

The brain uses low-frequency sound to accurately gauge the apparent size of the acoustic space that you are recording in. With that in mind, it makes sense to try to reduce the amount of low-frequency noise as much as possible to avoid having to roll off low frequencies during tracking or mixing. So, turn off large appliances such as refrigerators, air-conditioners, furnaces, and other large, vibration-causing machines before recording anything.
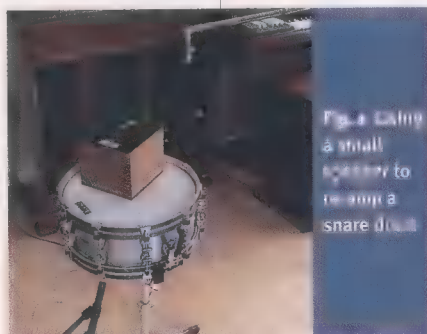
Catch Me If You Can

Though it sometimes seems that, as Chris Isaak said, it's a heart-shaped world, there's a reason why microphones have patterns other than cardioid. Take a look at the polar pattern sheet that came with one of your multipattern mics. (I'll wait right here. Good.) Notice that the frequency response varies with the selected pattern. This is simple physics.

Skipping the "why" for now, it's good to remember that omnidirectional mode will almost always yield a deeper and truer low-frequency response than the ubiquitous cardioid mode. If extended low frequencies are a good thing, then why do people use cardioid mode so often? Mainly, convenience. It is much easier to get a decent-sounding result quickly with cardioid mode, since it is less sensitive to room sound and room placement. Cardioid mode is also much more tolerant of rotten-sounding rooms and well suited to close-miking. In addition, most cardioid microphones produce a pleasing low-frequency augmentation when used in a close-miking application.

Microphone positioning is a mixture of art and science, with a small dollop of luck thrown in for good measure. The most important tools are, of course, your ears. Actually, let's make that your *ear*. Most people find it's much easier to hear the effects of the room when listening with just one ear. This is due to a mechanism of our hearing that allows the brain to cancel out the effects of room ambience for clarity's sake. Listening with just one ear will usually yield a much better sonic image of the overall room ambience. I often find it helpful to walk around the room with one ear covered when I am searching for the "sweet spot" to place ambient microphones. This technique also works very well for finding the optimum place to position close microphones.

When positioning cardioid microphones (especially for ambient pickup), you will often want to place the microphone so that it is pointed at the sound source itself. You are guaranteed to pick up an entirely different sonic picture with the microphone if you orient it away from the sound source that you are recording. Orienting the microphone away from the sound source will provide you with a much more ambient presentation of the instrument. As mentioned earlier, pointing (and positioning) the microphone in a corner will yield a greater low-frequency response.

Microphone Positioning Arrangements

ORTF A technique adopted by French Radio in which two cardioid microphones are positioned 17 cm (roughly the same as ear spacing) apart and angled outward at 110 degrees. This method can provide a greater sense of space than coincident techniques due to the microphones being ear spaced and thus capturing phase information as well as intensity information.

X/Y (Blumlein Pair) In this arrangement, a figure-8 microphone is placed on top of another to make the capsules as coincident as possible, then angled at 90 degrees to each other. While this produces a realistic soundstage, individual instruments (images) sometimes seem a bit small.

M/S (mid/side) This technique also uses two microphones, one facing forward (the mid mic) and the other facing to the side at right angles to the first mic (the side mic). The mid mic can be any polar pattern, but the side mic must be a figure-8. The output of the two mics is then fed into an M/S decoder. This converts M/S into standard left/right stereo by sum and differencing the two M/S signals. Some preamps now feature M/S matrixing.

Since omnidirectional microphones pick up sound (at least on paper) from all directions equally, you will want to make sure that you listen to the sweet spot that you have located (using the single-ear technique) with both ears in order to get a better picture of what the microphone will be picking up. I also like to spin around in place to make sure that I am hearing the different reflections that exist at that particular position in the room. This becomes especially critical when using a microphone pattern such as figure-8, or any of the other intermediate patterns that exist between the cardioid and omnidirectional settings. Those settings, by the way, are often very handy when the overall sound is good but a particular aspect (be it leakage from another instrument or an undesirable reflection) is causing an anomaly in the sonic picture.

When miking in stereo, there are a few common positioning arrangements that can be used. See the sidebar "Microphone Positioning Arrangements" for further explanation.

Reanimator

Oftentimes, especially when mixing someone else's project, you may not be happy with the tracks as recorded. One way to punch up the excitement is to *re-amp* the tracks. Contrary to popular belief, you don't need any specialized boxes to do this (though products such as the Radial JD-7 and the PCP Labs Distro can make it easier). Re-amping simply entails taking selective elements of the mix (and occasionally entire mixes), sending them to a loudspeaker, and miking the results. A good-quality condenser mic will make your job easier, but it should not be regarded as an essential requirement of re-amping. I have achieved excellent results on occasion with the ubiquitous Shure 57.

Re-amping works especially well for drums, allowing for a sense of the kit to develop, even from drum tracks that are dry and uninspiring. This can really improve a drum track that sounds too sterile due to the room in which it was originally recorded or the way it was recorded. Re-amping is also a lifesaver for animating electronic drums, as it creates a central location for all of the drums to originate from, just like an actual drum kit. During the mixing stage, by heavily compressing the re-amped tracks, a real sense of the room will emerge from a previously lifeless track.

Re-amping can also be a great technique for spicing up a bland-sounding snare drum. One neat trick involves sending the snare signal to a small speaker that has been laid on the top head of a snare drum and then miking the result (see **Fig. 1**). A microphone placed close to the snares of the drum adds some snap, and a more distant microphone will add a nice three-dimensional feel to the drum track. Once again, compression will be your good friend when it comes to dialing up the "size" factor on the re-amped tracks.

When re-amping, it's often helpful to use bone-dry tracks whenever possible, as re-amped ambience can sometimes produce some unpredictable results. Although I like to try and capture ambience in the original recording, I have often used ultra-close-miking when faced with a terrible tracking room or extreme-volume situations that preclude the proper isolation between instruments, rather than record tracks that will contain troublesome ambience or leakage.

Clients stay glued to the seat longer at Syn Studios.

The on-screen action is too good to miss.

"Making music with Logic Platinum 6 is an exciting visual experience, so clients like to have it on their personal monitors to see what we're doing." says Nick Wood, founder of Tokyo's Syn Studios. With its adaptable screen architecture that lets users define their own flexible workspace, Logic provides more time to enjoy the musical action on screen. "When I'm working on a commercial with Simon Le Bon of Duran Duran or a record with Chris Corner of the Sneaker Pimps, everyone's individual Logic 6 setups can be easily brought into the studio," adds Nick. "No need to carry computers from London or New York, clients just bring the data, ready to go." It's a pleasure to focus on the creative, not the technical, and reconfiguring for each session has never been easier. New features in Version 6 take these concepts even further. The Video Thumbnail Track and DV Video Playback via FireWire open new creative possibilities for sound to picture projects. The Marquee Tool, Time Stretching and Sample-Accurate Display all make composing music and editing audio directly in the Arrange window more convenient than ever before. Not only than, but the Project Manager adds a whole new dimension of efficiency when working with multiple clients. Jump out of your seat and get to your authorized Emagic dealer today, to check out Logic Platinum 6 for Mac OS X.

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Logic Platinum 6

The First Mac-Only Version Is Insanely Great

NOW THAT LOGIC (SEE "THE KILLER APPS, Part 3" page 42) is a Mac-only application following Apple's acquisition of Emagic, longtime Logic users no longer need to wait to see if the development for a single platform—as well as support from Apple—will have an effect on the speed and power of Logic upgrades. The first major upgrade since the buyout seems to live up to users' expectations, with a wealth of features introduced and some longtime issues and user requests finally addressed. Logic 6 is the first Mac-only version of the program, and despite some anticipation to the contrary, it still supports Mac OS 9 as well as OS X.

TAKING CARE OF BUSINESS

Along with the new features, Logic 6 provides solutions to long-term user issues. One result is full-featured support for Propellerhead Software's ReWire 2 protocol. While fans of apps like Reason were able to pump audio from Reason's instruments into Logic's mixer, now they can record and play Reason instruments from within Logic as well, via new ReWire Environment Instruments. Another new feature, the sample-accurate display of audio waveforms in the Arrange window, eliminates a persistent graphic display bug involving tempo changes. The loss of support in OS X (and later versions of OS 9) for some of the most popular video cards had left Logic users without an inexpensive way to play embedded QuickTime movies on a separate video monitor. The introduction of support for streaming DV-format QuickTime video via FireWire has dealt with this issue, and Emagic has even added a video thumbnail track for good measure.

MIXING ENHANCEMENTS

For a while now, Logic users have set up a "floating" channel strip—a 1-channel-wide Track Mixer window that followed track selection in the Arrange window. Emagic has now incorporated this feature into the Arrange window itself. And for those poor souls without a moving-fader control surface, Logic 6 offers Automation Quick Access, which lets users assign a hardware slider or wheel to control the selected automation parameter in any track, as opposed to drawing in mix moves with the mouse. Another new mixing enhancement is Grouping (a feature on many Logicians' wish lists). Logic now allows for grouped mixer channels to be controlled by any fader in the group, preserving the relative balance between levels. Groups can control not only volume, pan, mute, and send levels, but also edit selection, zoom, record-enable, and automation mode. A key command easily disables Grouping to permit adjustment of single faders.

My favorite new mix feature is the Channel EQ plug-in, an 8-band EQ with low- and high-cut filters, low- and high-frequency shelving, and four bands of boost/cut, all with adjustable frequency and bandwidth. Channel EQ also incorporates a built-in (optional) FFT Analyzer, which displays the spectral content of the signal being processed, to help home in on problem areas. Best of all, the EQ sounds smooth, warm, and transparent—up there with high-end third-party plug-ins. And it's more CPU-efficient than its predecessor!

AUDIO PROCESSING

Logic 6 also offers a couple of nice audio improvements. A welcome addition to the Bounce feature is the ability to perform offline (faster

FEATURES

- 32-bit internal processing
- MIDI event resolution to 1/3,840th of a note
- Tempo resolution accurate to 1/10,000 bpm in the range from 0.05 to 9,999 bpm
- Freely configurable user interface
- 90 freely configurable Screen Sets per song
- More than 800 definable key and MIDI commands for all important program functions
- Individually zoomable/resizable tracks
- Synchronized QuickTime video support

System Requirements

Processor:	Power Mac 604/250 MHz or better
RAM:	128 MB or more
OS:	Mac OS 9.1 or higher/Mac OS X 10.2 or higher
Other:	Free USB port for XSKey

By Joe Albano

A screen shot of Logic 6 showing the video thumbnail track, the Channel EQ thumbnail (in the channel strip), the Group Settings dialog box, a Marquee selection in the Arrange window, and an assortment of new high-res icons in the track list.
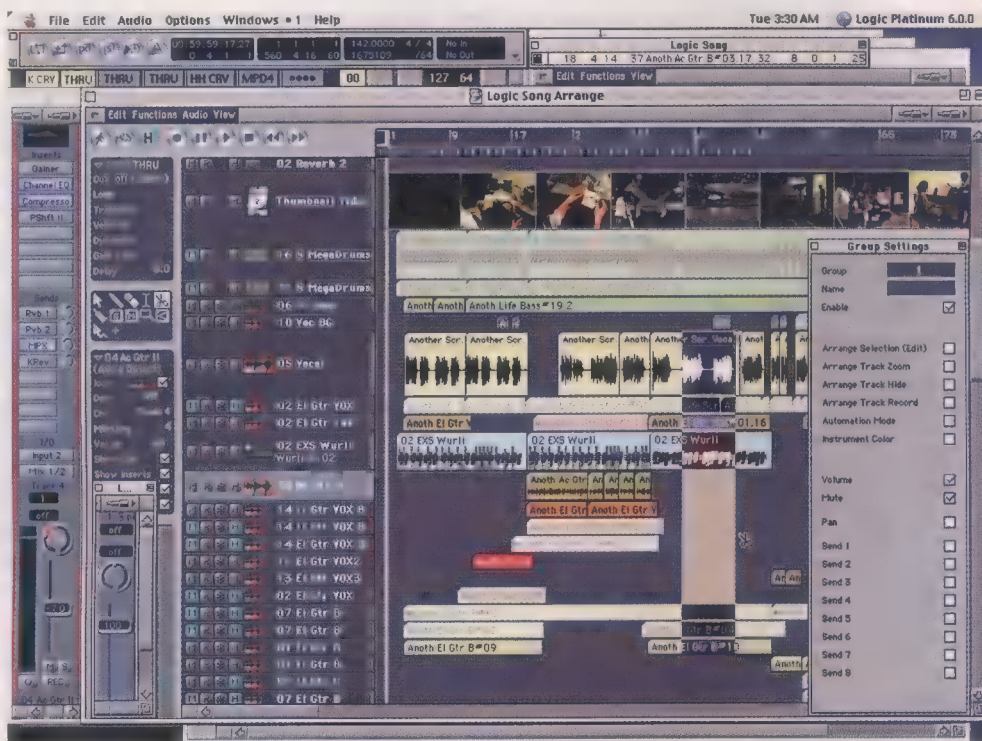
than real-time) bounces, another much-requested option. And Logic's Time Machine, which has been around for a long time, has been beefed up with five new high-quality, high-accuracy pitch-shifting/time-stretching algorithms. Not only that, but it's now possible to perform some time stretching right within the Arrange window (without having to click into the Sample Editor). You can squeeze or stretch an audio region to conform to a selected range of bars, or you can force the end of the region to line up with the nearest bar; this is a convenient way to quickly match loops to a song's tempo.

ARRANGE WINDOW GOODIES

A new tool has been added to the Arrange window's toolbox: the Marquee tool. Moving and cutting up sequences in Logic had previously been done with the arrow (pointer) and the scissors tool. The new Marquee tool (resembling a crosshair) adds a method familiar to users of Pro Tools and other DAWs. Dragging inside an existing region will define a new shaded region that can then be edited further. Marquee selections can also be made across several existing regions at once, and these can all be edited together. For particular types of edits that required several steps with the previous method, the Marquee tool offers a speedy alternative, especially for users who

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$949

move between different workstations and are used to this technique from other programs.

Another convenience feature is the Hide Tracks function. Hide buttons have been added to all tracks and work in conjunction with a master Hide button in the transport area. Tracks whose Hide buttons are enabled will, naturally, be hidden from view when the master Hide switch is thrown—they can even continue to play while hidden. Okay, so that's pretty obvious, but it's still a great tool for keeping the main work area clean and uncluttered. Finally, to spruce up the track list, those tired old track icons that have been around since day one have been slicked up with big, bold, colorful new ones—you can even roll your own (in Photoshop or the like) and import them into Logic.

FOR OS X ONLY

A couple of features are reserved strictly for those who've made the jump to OS X. One is the Setup Assistant, which simplifies the initial setup of Logic for newbies. The other is support for MP3 import (conversion) and export, the latter handled in the Bounce dialog box, with a full range of quality/size settings available.

KEEPING IT ALL TOGETHER

With the plethora of files that go into a typical Logic song these days, it's a wonder that anyone can keep track of them all. Even a well-organized song file may have not only dependent audio files,
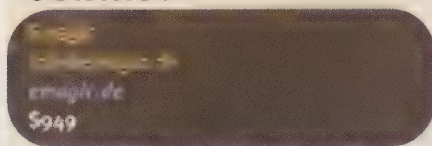
but also Sampler Instrument (program) files, the actual audio samples that go with them, plug-in settings, and sometimes QuickTime movies, all (except for the audio files) stored in different locations outside the song folder. Enter the Project Manager, a comprehensive database within Logic that searches the user's system and resolves the relationships between all Logic songs and their dependent files on various disks. There's simply too much to this feature to delve into all the details here; in a nutshell, the PM provides intelligent file management, assists in creating backups and archives, and promises to give busy Logic users a leg up in keeping sessions and projects well organized.

FROZEN DELIGHT

I've saved the best (at least in my opinion) for last. There's one more new Logic feature that's gotten a lot of attention: Freeze. The Freeze feature is deceptively simple in concept but tremendously useful in practice. No matter how powerful computers get, we always seem to push them to their limits, especially when mixing. There are always one or two great-sounding but CPU-hungry plug-ins or soft synth patches that manage to elicit the dreaded "CPU Overload" message at just the wrong time. Freeze is designed to reduce CPU load when the computer is maxing out from tracks with plug-ins or virtual instruments that use a lot of processing power. Freezing a track bounces it down into an audio file, preserving all the CPU-intensive effects and polyphony but using only a tiny amount of CPU power for audio playback of the new Freeze file. The volume, pan, sends, and other mix controls can still be adjusted, but the instrument and plug-in settings are fixed in the Frozen file. If adjustments must be made to Frozen parameters, simply turning Freeze off for that track restores the original audio file and live plug-ins; the adjustments can then be made and the file re-Frozen with the changes. Of course, this is something that could be done manually, but keeping track of temporary bounces and restoring the original versions for edits can be so cumbersome as to be impractical; the Freeze feature reduces this to a couple of clicks, taking only seconds yet still allowing for basic mixing of the track even while Frozen. This has proved invaluable on more than a few mixes I've seen that were just at the edge of CPU overload, giving new life to several systems that were struggling. This feature is a real winner!

THE LAST WORD

Just as this article was about to be submitted, Emagic released the first Logic 6 subrelease, version 6.1. The main features include added support for TDM systems under OS X, implementation of Akai conversion in the EXS sampler on OS X, and various enhancements to the Project Manager, Setup Assistant, and control surface support, as well as minor tweaks and improvements throughout the program.

THE VERDICT

At the beginning of this review, I mentioned that Logic users had been waiting for this update to see how the changes at Emagic would affect development of their favorite program. Well, by now, it should be obvious that Logic, at least on the Mac, is going full steam ahead. If the version 6 upgrade is an indication of things to come under the new Apple/Emagic team, then the future looks bright indeed for this powerful application. ▌▌▌▌

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Zoom
MRS-802CD
Fast Work With a Workstation Wonder

COMPUTER-BASED OR STANDALONE? That is the question when choosing a recording system. Software systems offer the advantage of a wide range of plug-ins and expansion capabilities, whereas dedicated stand-alone units remove conflict worries while providing portability and convenience. The Zoom MRS-802CD exemplifies the latter, allowing you to take a project from recording to mastering to burning—all in one box.

The MRS-802CD features a 20 GB hard drive that provides roughly 60 hours of 16-bit uncompressed recording time. It boasts a 44.1 kHz sampling rate and 24-bit A/D/A signal processing. The recorder features eight mono digital recording tracks, a separate stereo track for the internal drum sequencer, and a stereo master track for mixing down, so it is actually like having a 12-track recorder. Each mono track and the stereo master track allow 10 virtual takes per track, enabling the user to pick the best one during mixing and/or save 10 different mixdowns before committing one to CD.

The MRS-802CD has 2-track simultaneous input capability, and two mono tracks can be linked to make stereo channels. It has 1/4" and XLR inputs (balanced and unbalanced) with 48V phantom power, as well as MIDI in and out. A range of nonlinear editing capabilities is available, including cut and paste, copy, move, reverse-direction play, and time stretch/compress. You can also import and edit samples and loops using the built-in CD-R/RW or the optional USB interface. Each track has editable parameters, including high and low EQ, pan, chorus, delay, and reverb. These are automated, allowing changes on any given track midsong. Parameter settings are saved as "scenes"—up to 100 scenes per song/project. You can save up to 100 musical phrases (either recorded or imported) in a Phrase Pool for loop composing.

The drum section features more than 400 drum and percussion

FEATURES

- 8 digital recordable tracks
- 10 virtual takes per track
- Drum machine
- Extensive bank of effects
- CD-R/RW included

sounds, including 43 different drum kits, each with 24 different sounds. The drum section is also fully programmable, allowing you to build parts from the ground up. Techno fans will appreciate that you can record the drums to the audio tracks through some cool, nasty effects. The MRS-802CD includes 280 preprogrammed and editable effects patches, available at the inputs. These are divided into algorithms for Clean, Distortion, Acoustic and Bass Simulation, Bass, Mic, Dual Mic, and Line. There are also 40 send/return effects patches for mixing and mastering. When mixed, a song or project can be mastered to the stereo master channel, then burned to disc with the CD-R/RW drive. If several projects are completed, an entire album can be sequenced and copied to a CD-R. Project data can also be stored to disc. Audio (16-bit) can be loaded into the unit from the CD drive, but only as a full track; trimming must be done within the Zoom unit after loading. The optional UIB-01/UIB-02 board allows you to connect the MRS-802CD to a Macintosh or Windows computer for audio data editing, backup, etc. (It does not presently support Mac OS X.)

Being a Pro Tools user, I was new to stand-alone recording systems. Fortunately, the MRS-802CD proved extremely simple to operate, thanks to a well-written manual. Going from unpacking to finished song idea took me only a couple of hours, with no phone calls to tech support necessary. I turned on the unit, found a drum groove, and went to work. Using the acoustic simulator effect, I put down some strummed chords with my electric guitar. Next, I ran the same instrument through the Stones-like Sticky patch for some Keith riffing. The Bass simulator proved to be a glorified octave device, serviceable if no real bass instrument is available. The funky filtered setting was pretty good, but the roots-rock concept I was working on required something that sounded more authentic, so I plugged my Danelectro bass in through the bass effect section. Not being a singer, I tested the mic

preamps by recording a Goldtone six-string banjo. The phantom power juiced up the Soundelux R-1 mic just fine, but I decided to go for a darker sound and plugged in a Shure SM57. One tremoloed melody line later, I was ready to mix. The Track Parameter button allowed me to roll unwanted bass off the "acoustic track" and pan it hard right; pan the banjo right of center; pan the electric riff to the left; add a little chorus to the melody line; and cut the highs on the bass. The send/return effects allowed me to send different amounts of reverb to different tracks. Using the Live mastering preset, I mixed down the song and burned it to the CD-R drive.

Having experienced the sonic improvement from adding even an inexpensive tube preamp and converter to my Digi 001, I wish that the MRS-802CD had digital inputs to allow the use of a higher-end pre-amp. Nevertheless, the important thing is that learning the unit did not hamper my creative impulse, and even this quickly realized sketch was one I would be proud to play for anyone as an accurate representation of my musical inspiration. ▌▌▌▌

CONTACT

Roland
VariOS

Powerful Hardware/Software Combo for Your MIDI Desktop

ROLAND'S VARIPHRASE TECHNOLOGY revolutionized musicmaking when the company introduced the VP 9000, allowing samples to be tuned, timed, and morphed quickly and easily with just a twist of a knob. Building on that platform, Roland has introduced the VariOS, a one-space rackmount unit that offers the flexibility of a dedicated module, with open architecture that allows expansion modules to be downloaded from the Internet right into your unit. Imagine a sample player with completely variable pitch, time, and formant controls, a drag-and-drop multitrack audio sequencer with onboard effects, a MIDI interface, and software-controlled synths that don't use up any precious CPU power, and you get the idea of VariOS.

EASY CONNECT

The attractive metallic-red one-space unit connects easily to your system. The rear panel sports a pair of 1/4" main audio outputs for monitoring and a pair of direct outs for dry, unprocessed sound or effects only. Stereo audio inputs for sampling; an input level control; MIDI in, out, and through; and S/PDIF and optical digital outputs complete the rear panel. The front panel has dedicated knobs for pitch, time, and formant editing, and menu navigation buttons to get around the menu displayed on the backlit LCD. The unit connects to the computer via USB and functions as a MIDI interface for your computer. Currently, the software runs on Mac OS 9.2 (which I was running) and Windows.

VariOS receives MIDI directly via USB and is configured like a standard USB interface. You can connect a keyboard or other modules to it, but it is controlled by Roland's V-Producer software, which allows sample loading, editing, and sequencing right from the desktop. VariOS "sees" three ports: internal (or USB), external (connecting the rear-panel MIDI jacks), and sync to clock for other hardware and software. After installing the driver and configuring OMS, I was off and running.

FEATURES

- Up to 14 phrases of simultaneous playback; 6-part multitimbral operation
- Real-time manipulation of pitch, time, and formant
- Internal memory holds up to 300 seconds of mono samples, 150 seconds of stereo samples
- Graphic editing of VariOS module's 40 multi-effects, 8 chorus types, and 9 reverbs
- Flash memory and PC card sample storage

V-PRODUCER

V-Producer is an easy-to-operate, six-part multitrack sequencer that supports drag-and-drop editing on a timeline display. VariOS can hold up to 14 different sampled phrases simultaneously. Samples can be browsed and imported from your hard drive, and as they are sent to the VariOS unit via USB, they are "encoded" as one of three types of sample: *Solo* encoding refers to monophonic melodic parts; *Backing* is for beat-oriented samples with substantial transients; and *Ensemble* encodes harmonic content like pads or background vocal harmonies. Once the samples are encoded, they can be manipulated by Roland's VariPhrase processing. This algorithm allows real-time changes to pitch, time, and formant with virtually no artifacts. Since the processing is taking place inside the VariOS engine, not your CPU, there is no latency and no system drain. Once the samples are transferred, they can be arranged in the sequence window with drag-and-drop editing commands or played in via MIDI notes. Any batch of samples can work together, because VariOS allows independent control over each phrase's pitch and time.

The main screen of V-Producer shows the timeline window for the six "parts" and a sample list showing all the samples currently in the unit's memory. VariOS supports .wav and .aiff audio files for encoding and can hold up to 300 seconds of mono audio (150 seconds of stereo). Next to the sample list is a sample wave editor that reflects the current settings of a selected sample. I found the sample editor intuitive and the icons for each parameter clear. The usual key assign, tuning, pan, and gain controls are coupled with formant, pitch, and time parameters unique to VariPhrase encoding. Sample playback occurs in two modes: the usual "retrigger," where each MIDI note plays the sample from its beginning at the pitch of the note, and "time sync," where

By David Darlington

subsequent MIDI triggers play the same sample synchronized to the beginning of the first retrigger. In this mode, you could reharmonize a sample of "Happy Birthday" to sound like "The Star-Spangled Banner." One interesting switch marked "Robot" flipped the playback into a vocoder mode that sounded very modern. Portamento and LFO controls round out the sample playback editing.

The Wave Edit page of the sample editor shows the VariPhrase encoding on a grid, with each "slice" of the wave marked by bar and beat. In this window, it is easy to rearrange the slices to fine-tune the encoding or requantize the feel of the sample. You can even slide the time grid to fit the sample and then re-encode for accurate timing of the sample playback. Samples are mapped for either phrase (pitched) or groove (timed) playback.

Once the waves are encoded and mapped, the fun really begins. The Scope Editor allows nondestructive editing of the two types of maps. The Phrase Scope Editor shows a graph of the selected wave over time, with its corresponding MIDI pitches below and a controller's grid at the bottom showing variations in pitch, time, formant, and dynamics. Think of it as a MIDI editor for audio files. Using the usual drawing tools (line, curve, flat, etc.), you actually draw changes into the pitch and duration of the sample playback. Changes are monitored back in real time, so you can fine-tune as you work. Rearrange a melody or lengthen a note, all nondestructively in real time. Remember that even though you're editing on your screen, the action is really in the VariOS module, so there is no tax on your CPU.

The Groove Scope Editor is the now familiar drum slices over time grid display. This window makes it easy to use slices of drum loops in unexpected ways, straighten out timing problems, or just repeat the same snare for every backbeat. Slices are numbered according to their original order, so you can't really get too lost. Just drag a slice or Option-drag to copy it to a new location. Slices can be time stretched or pitch shifted right from the screen, or if you prefer, highlight them and grab the knobs on the front

panel. There is even a randomizing feature to stir up your beat and create new grooves from existing material.

V-Producer syncs easily to other sequencers like Performer or Logic via MTC and MIDI clock, and can export arrangements as .smf or encoded audio files. Entire arrangements can be saved as a stereo .aiff or .wav file. Samples can be saved to flash memory or PC cards, so the unit can be played via MIDI without the computer.

OPEN SYSTEM SYNTH MODULES

On the Roland Web site, owners can download synth playback modules that use the VariOS playback engine and are controlled by displays on the PC. Currently available are a VariOS 303 and the very analog-sounding VariOS-8. The downloaded programs are dragged to the flash ROM via USB and enable the VariOS to function as a stand-alone synth module. After a two-minute download and a one-minute install, I had two new vintage synths in my arsenal. The controller programs appear on the desktop, giving all the convenience of a virtual synth but without the CPU drain. Listening to these synths via the S/PDIF output into my DAW, I was truly amazed. They sound great! The VariOS-8 sounds like both the revered Jupiter 8 and the old faithful Juno 106. The VariOS 303 sounds and sequences like that techno beauty the TB-303. Cost of the additional synth modules: $0. Thank you, Roland!

The VariPhrase architecture makes it fun and easy to juxtapose loops and samples and make music in new and exciting ways. VariOS allows you to reharmonize, time stretch, and arrange on the fly nondestructively. The convenience of the V-Producer software allows you to experiment and keep track of all your resources easily. Add great synths (with more to come), and you have a formidable creation tool. I can't wait to see what Roland designs next for the Open System, but one thing I am sure of: It will sound tremendous.

CONTACT

Roland

[unreadable contact details]

ART
Tube Preamp System

A Few New Twists in a Sonic Toolbox Bargain

WHO SAYS YOU CAN'T TEACH AN OLD DOG new tricks? Reviewing a mic preamp is generally a less-than-thrilling assignment, truth be told. Sure, every preamp has a characteristic sound quality, but there's rarely more than a pair of input/output gain controls to tweak, along with the obligatory phantom power switch, phase reverse switch, and perhaps one or two other bells and whistles.

Enter the ART Tube Preamp System (TPS), a 2-channel tube mic pre-amp that adds a new twist that will make Rover (assuming you have a microphone named Rover) sit up and beg for more. This is accomplished by a unique control called V3 (Variable Valve Voicing), which provides 16 presets optimized for recording vocals, bass, electric and acoustic guitar, piano, electronic keyboard, and percussion. Each preset adds a different amount of tube saturation and output limiting so as to best enhance the sound of the particular signal source. Of course, being a contrary guy, I couldn't help but experiment with using presets that were optimized for signals other than the ones I was recording—and interestingly, I often came up with excellent results!

In effect, the V3 control gives you 16 different timbral opportunities every time you plug a mic or instrument into the TPS (which can also be used as a DI box), making it a kind of sonic toolbox instead of a one-trick pony. (Sorry I'm mixing animal metaphors, but you get the point.) At a list price of just over $100 a channel, that's nothing short of amazing.

But wait—there's more: Both balanced and unbalanced input and output connectors are provided (in both XLR and 1/4" configurations; there are even dual-purpose jacks on the front panel, which can accommodate both kinds of plugs), as well as independent VU metering for each channel to show you how hard you're hitting the 12AX7 dual-triode tube. (The more level that is sent to the tube, the greater the amount of tube saturation, or warmth, is imparted to the incoming signal.) More than 75 dB of gain is provided, with a handy +20 dB switch that allows you to boost weak signal, enabling

you to use the TPS with a variety of microphones and instruments. This switch, along with a continuously variable input control, is the key to getting the most out of the TPS, since it determines how much tube saturation is added. Fortunately, the VU meters allow you to adjust this with great accuracy so that, with careful tweaking, you can maximize the amount of tube effect while minimizing the amount of output signal limiting. By the way, the always in-line OPL (Output Protection Limiter) does a great job of eliminating clipping—this box is damn near impossible to overload!

Even though in the audio world you sometimes get only what you pay for, I try not to allow pricing to affect my thinking about a product. But I will admit that I was not expecting to be blown away by a $219 mic preamp, especially when compared with other, considerably more expensive preamps in my home studio. I was, however, enormously impressed with the sound quality of the TPS both as a vocal mic pre and in use as a DI box. I tested it with an Audio-Technica 4050 large-diaphragm condenser mic—recording both male and female vocals, as well as acoustic guitar using a variety of V3 presets—and it shone in all three applications, in some cases even outperforming the aforementioned more expensive pre's. Since bass is my main instrument, I was especially anxious to try the TPS as a bass DI with my brand-new Aria TAB-60 semi-hollowbody bass, and was floored by the rich, creamy sound it imparted. It did a less spectacular job as an electric guitar DI, however—none of the presets really brought out the best in either of the electrics I tried, though this may well be an inherent limitation of using a tube device with an instrument that requires extremely fast transient response.

True, the TPS is not the most neutral-sounding mic preamp I have ever heard, even when the Neutral Flat preset is dialed in; it delivers a slightly dark sound, with a tendency to emphasize low midrange frequencies. But in practice, one or more of the presets almost always served to color the input signal in a pleasing way, instantly making the TPS one of the most useful devices in my studio. I highly recommend it! ▮▮▮▮

FEATURES

- V3—Variable Valve Voicing
- OPL—Output Protection Limiter
- 5 Hz–150 kHz frequency response
- Combination XLR–1/4" front panel inputs
- +20 dBu max. input level; +48V phantom power
- Phase reverse switch on each channel

KRK
E8T

An Exposé on Great Monitors

ET'S GET RIGHT TO THE CENTRAL QUESTION. Who would pay $5,000 for a pair of nearfield monitors? Plenty of people, it turns out—people whose home studios are ready for the big leap into semi-pro land or who designed their project studios to compete with the big boys from the get-go. We all know that more and more major-label releases are coming out of home-based studios, and if you're approaching that level, it only makes sense to apportion a hefty chunk of your equipment budget to monitors that stand with the best in terms of design, sturdiness, and long-term performance as well as sound quality. Assuming that the KRK E8Ts—the newest members of the company's Exposé series—live up to those requirements, five grand may be a small price to pay for the goodwill and reputation these monitors could help your studio attain. (Skeptical? Well, you can find pairs of E8Ts selling in the neighborhood of $3,400 at several online pro audio dealers. That's less than a top-of-the-line synth or a state-of-the-art computer cost back in the day, and your monitors certainly work as hard as those studio essentials.)

If all this seems like putting the cart before the horse, forgive me. I'm a KRK fan from way back. I was first exposed (no pun intended) to the company's products by producer Rob Stevens on a Yoko Ono session almost 10 years ago. The crisp clarity of his KRK 9000s brought immediate and welcome relief from the incessant NS-10 nearfield nightmare. But though the company has changed locations and ownership in subsequent years, the quality of their products has remained high. In short, I've never met a KRK I didn't like. Ultimately, for me, like every other home studio owner, the question gets down to price.

What makes the E8Ts so great? The components, for one thing. Two symmetric discreet power amplifiers provide 160 and 130 watts to the woofer and tweeter, respectively. Included is newly designed crossover circuitry. A double-woven Kevlar woofer reproduces accurate and hefty bass without muddiness. The tweeters use Tioxid 5, a material that mixes titanium with a coat of titanium dioxide. Along with the unique inverted-dome shape of the tweeter, the design helps create a more neutral high-frequency response.

This is borne out by several listening tests. The 8" Kevlars, marked by KRK's distinctive yellow finish on the cones and aided by a 12" slotted port at the bottom of the front panel, pump out way more bass than you'll ever need—even mixing club music in a medium-size room. But the bass is always well defined. Even at low volumes, the E8Ts maintain their transparent crispness, and the sound is uniformly great on everything from delicate guitar/vocal passages to pounding sample-based hip-hop. The E8Ts even clearly showed a flaw in another set of powered monitors with 8" drivers, about which I'd recently raved in these pages. A low-volume edgy guitar turned out to be a relatively clean instrument when monitored on the E8Ts. The other monitors, a foreign-built set that otherwise sounded great, were actually creating almost inaudible distortion on the left side when reproducing certain midrange frequencies. It was a telltale flaw I'd missed in my review. The KRKs exposed it (there's that word again) for what it was. (In fairness, the flawed monitor cost almost $1,000 less than an E8T.)

The E8Ts are beautiful in their simplicity: 61 pounds of hexagonal cabinet putting out 280 watts of powerful and pristine sound, with no sub to hook up, one combi connector for a balanced XLR or 1/4" cable, and a power cable and switch on the rear panel. That's it. The only disappointment: At this price, the power indicator should be mountable on the front panel instead of the rear, where it accomplishes nothing after the initial setup. But this is a small blemish indeed for one of the best-sounding, sturdiest, and, yes, value-priced monitors on the market. ▮▮▮▮

FEATURES

- **Standard video shielding**
- **HF 1" Tioxid 5 inverted dome, LF 8" Polykevlar drivers**
- **HF 120W @ 8 ohms, LF 160W @ 8 ohms**
- **Neutrik XLR 10 kΩ balanced input**
- **Frequency response: 42 Hz–21 kHz, ±1 dB**
- **Maximum SPL @ 1m: 110 dB music, 113 dB peak**
- **17½" (h) × 14¹⁵⁄₁₆" (w) × 14½" (d)**
- **61 lbs. each**

CONTACT

KRK
805-484-6484
krksys.com
source list

Two Sets, Rad and Trad

Caged Piano Sings, Celtic Instruments Ring

Big Fish Audio
JOHN CAGE PREPARED PIANO

Many musicians know that John Cage experimented with altering the sound of the piano to compose and perform his pieces. In the past, pianists had to laboriously prepare an instrument for Cage's works. This involved inserting various pieces of rubber, metal screws, bolts, and nuts between the strings of specified notes on the soundboard at precise distances from the damper or bridge. Now Big Fish Audio has released *John Cage Prepared Piano* ($199.95), a CD-ROM sample set of a piano prepared according to Cage's instructions, to a large extent using his original materials. The library comes in Akai S1000 format and in GigaSampler/Halion format, which is the format I used for this review.
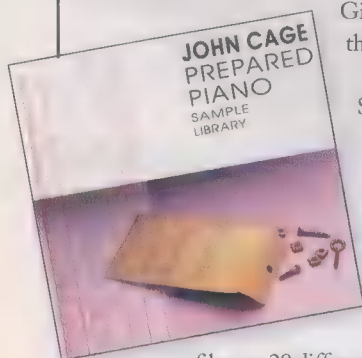
There are four .gig files: Cage Piano ST, Cage Piano M, Cage Soft Pedal ST, and Cage Soft Pedal M, with ST and M denoting stereo and mono. The Piano samples were recorded normally (if that's the right word for it), and the Soft Pedal samples were recorded with the piano's soft pedal down. Within each of the four .gig files are 29 different instruments, most of which are variations of prepared piano setups, with different MIDI parameters controlling performance combinations or aspects. For example, Cage SusSw plays naturally decaying sounds when the sustain pedal is down, but staccato samples when the pedal is up.

The sound itself has a randomly produced synthetic texture that sounds oddly natural. For all its alterations, it's still an acoustic instrument. Obviously, you can use this library to play Cage's works, but why stop there? The unique sound begs to be used in a variety of musical contexts, from electronica to film noir soundtracks. Couple it with GigaStudio's effects (or VST effects if you're a Halion user), and you have the makings of some interesting sound textures. *bigfishaudio.com*

Big Fish Audio
CELTIC INSTRUMENTS

This CD-ROM set ($299.95) in Akai S1000 format and GigaSampler/Halion format (the one reviewed here) contains .gig files of 13 Celtic instruments, including Bodhran, Button Box Accordion, Celtic Harp, Fiddle, Low Whistle, Pennywhistle, North Umbrian Small Pipes, Ullian Pipes, and others. Within these files, there are Instrument patches that contain variations of playing styles for most of these sounds. For example, the Bodhran variations include Hits 1 and 2 and Hits 1 and 2 Filter (which use velocity to filter the sounds). Similarly, some of the Fiddle instrument variations include Fiddle Slides and Fiddle Turns. As you may guess, Slides are samples of up and down fiddle note slides, while Turns are phrases that can lead from one musical passage to another. These variations are best used in combination with a fiddle instrument suited for soloing, such as Fiddle Slow Attack Down.

I like this collection overall, but am not totally happy with the *Celtic Instruments* library. Sample libraries like this often group so many ethnic instruments together that some instruments don't get the attention they deserve. For instance, the Button Box Accordion doesn't have a corresponding set of chord accompaniment samples to go with its single-note samples. Likewise, I wish the Guitar also had a set of major, minor, and dominant 7th strum chord samples to use with the single notes of the Guitar instruments.

On the positive side, the collection of all four pipe instruments is outstanding, and the fiddle, harp, and whistle samples are solid, useful instruments. I just like what's offered in the *Celtic Instruments* collection so much that it whets my appetite for even better sample libraries based on some of its sounds (like a complete fiddle library). Even so, it's a useful library, and worth recommending. *bigfishaudio.com*

Ambient Recording

The goal to keep in mind when re-amping is to find the appropriate acoustic space to use as the re-amping environment. Often household (and studio) spaces such as bathrooms, attics, storage areas, stairwells, and hallways can yield excellent results that would be nearly impossible to model digitally. Another option is to use your tracking room as your re-amping space. This can be a godsend when trying to match the sound of an original instrument while overdubbing.

Another method is to mic your control room and re-amp your entire mix as it's played back through the studio monitors (see **Fig. 2**). But be careful when you do this, as a nasty feedback loop could occur if you have your gain set too high. Send the output of the microphones to tape, and mute the channels that they return to. Later you can add the re-amped sound to the rest of the mix.

If you really want to get radical, you can take a page from the Tchad Blake school of ambient trickery and use common objects as containers for your microphone. Mr. Blake is well known for putting microphones into boxes, bags, pots and pans, instruments, and other enclosed spaces to alter their frequency response and resonant behavior. Be sure to add lots of grainy compression to complete the effect.

Hear and Now

Now that we've captured some good-sounding ambient tracks, it's time to mix them in to the recording. Perhaps it runs contrary to intuition, but ambient tracks are often better in mono than stereo, especially when you want to call attention to them within the mix. I often like to take an electric guitar solo recorded ambiently and pan it right down the center. This gives the track the illusion of existing in a deep well of space, at the heart of the recording. Generally, this will have much more emotional impact compared to the same track panned either hard left or right. The track panned hard in either direction will tend to feel unbalanced, as the ear is looking for the complementary track on the other side. Of course, you can always double your track and pan the two to opposite sides!

Stereo ambient tracks will usually work best when they are panned hard left and right.

Fig. 2. Re-amping the entire mix by miking the control room monitors.

In the case of an ambient drum track, I like to bring up the ambient tracks first, and then add the close-miked kick and snare tracks (toms and overheads, if available) to add some impact and, if necessary, restore the proper balance. Sometimes well-recorded ambient drum tracks won't even need any spotlighting with the direct tracks, but this is rare.

Compression and gating will allow you to alter the apparent acoustic environment in ways that you might not be accustomed to if you have been working primarily with close-miked tracks. When you compress an ambient track heavily (with a fast attack, medium ratio, and fast release), you will increase the apparent ambient-to-dry mix. This technique will almost always make the room seem bigger than it actually is. Gating, on the other hand, can restore a sense of direct sound (dryness) to the room by cutting off the ambient tails of the track. I like to use a medium ratio of gating (in other words, the gate does not close down to silence; instead, it just reduces the level by a preset amount) to avoid what I call "Phil Collins–itis" (or obviously gated reverb effects, which have been out of style since approximately 1987).

If you are working on a DAW or digital tape system that allows you to delay selected tracks, you may want to experiment with moving the ambient tracks in time so that the transients line up with the close-miked tracks. Although it is within the realm of the physically impossible (in real space), it does eliminate the danger of phase cancellation between the dry and ambient tracks. Generally, when combining dry and ambient tracks (of the same source), you are going to want to flip the phase of one of the tracks to find the setting that yields the better low-frequency response. Keep in mind that this should be done *before* applying any equalization, as most EQ introduces phase anomalies.

Keeping It Real

For most styles of music, a dose of spatial reality really enhances the production. Most people find it easier to listen to tracks that have some ambient information embedded in them, as it allows them to discern a sense of "place" in the music. People are accustomed to hearing sound during the course of their lives, all of which takes place in ambient environments. Shouldn't your music as well? ▐▐▐▐
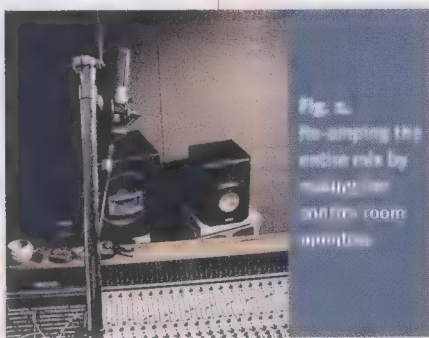
50 Guitar Tips

and effort to find the perfect NOS Mullard 12AT7, but often play with the same pick for a lifetime. Thinner picks can soften rhythm parts for ballads and change the timbre of lead lines. Really heavy picks, or even things like coins, give an abrasive and sharp sound.

Add way too much slapback echo. From rockabilly to prog, slapback just sounds good on guitars. Try short delays, a 16th note or less, on almost any guitar part. The more prominent the delay, the more it starts to sound like Sun Studios–era Elvis, which is not at all a bad thing.

Use fewer effects. Guitar players tend to layer on tons of effects live and when practicing. Effects that sound great onstage or in isolation, though, don't always sound good in a mix. Too many effects, particularly modulation effects, rob a guitar part of its impact.

Try compressing with a stompbox. Many people compress guitars when mixing, but in some situations, the compression sounds better when applied to the signal before it hits the amp. The MXR Dynacomp and the various boutique re-creations of the Boss compressor are fairly cheap ways to get some "pre-amp" compression.

Reps and Extras

STRING PICKUP. FOR THAT LITTLE extra shimmer on electric rhythm parts, try miking the guitar itself as well as the amp during the performance. Then add in a tiny amount of that plectrum-on-strings sound with the main amp signal to create a distinctive in-your-face track.

Squash it. For acoustic guitar with electric attitude, place a condenser microphone close to the sound hole, then run the signal through a compressor with the attack-release set to stun.

Make it two. Record an acoustic guitar part. Then copy it to an open track, dialing in a smidge of delay along the way. Pan the tracks hard left and right, and presto—two guitars for the price of one! ▐▐▐▐
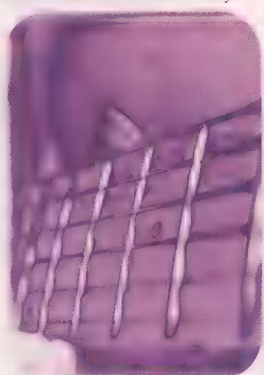
home Recording
Classifieds

What would you do if you were awash in microphones and could set up as many as you wanted?

RECORDING LIVE
INSTRUMENTS 101,
PART 6

MORE DRUMS, MULTIPLE MICS

YOU CAN GET A GOOD DRUM SOUND WITH just a single mic, but what would you do if you were awash in microphones and could set up as many as you wanted? Where would you put them all, and how many is too many? Here is a primer for those of you who are unabashedly spoiled with more than enough mics.

A great kick-drum recording requires the attack of the beater for high-end clarity and the resonance of the drum itself for low-end power. Unfortunately, these two sonic elements are not easily recorded in one position. I like to stick a dynamic mic (like an AKG D-112) inside the kick drum in the center, close to where the beater hits. This mic picks up the attack of the sound. Outside the hole, about a foot away, I place a large-diaphragm condenser mic to record the developing low frequencies. My favorite is a Neumann FET 47. I use a windscreen in this application to avoid damaging the capsule. Record each mic to its own channel, then blend them together for the perfect bass drum sound. Don't forget to check the phase coherency of the two signals.

Snare drums can also benefit from this two-mic approach. Of course, the stick work will be clearest and most present if you mic the top of the drum. A Shure SM57 or other dynamic mic works quite well. I try to position the mic 1–2" off the head, just inside the rim but aimed toward the center. You definitely need to move the mic around a bit while the drummer plays to get the best results. A player with a light touch can be miked closely, while a real banger may need a little distance to get that "big" sound. The other component of the snare sound comes from the strainer underneath the drum, which adds the "buzz" and decay tail to the sound as it rattles sympathetically to the top head. A high-quality dynamic positioned underneath and in the center of the drum will grab this sound. I prefer a Sennheiser MD 441 for this application. Once again, check the phase.

Hi-hats need to be crisp and clear, so I recommend a high-quality condenser for separate hat miking. Good choices here are the AKG 451 or 460 or Neumann KM 84. I like to mic the outer edge of the top hat, with the mic pointing straight down toward the cymbal. It must be high enough so that the open and closed sounds are balanced. If the mic is too close, the open sound will be too loud. Some engineers angle the mic in toward the bell a bit, but this will pick up more of the snare sound. Experiment to get a good blend.

Tom mics can be positioned close to the head and angled toward the center with a good dynamic mic like the AKG MD 421. The 421's pattern is very tight, so there's not a lot of bleed from the other drums. I've even heard of miking the underside of the toms in reverse phase to achieve a bigger tom sound, but to me, this seems like gluttony. Overhead, I like a matched pair of condensers like Neumann KM 84s or, for a warmer, more natural sound, my Royer 121 ribbons. I position the overheads left and right over the kit and focus on the cymbals. I don't mind if one mic is a little closer than another, as long as the cymbals sound well balanced. Some engineers like to use an X-Y stereo pattern; others don't use stereo overheads at all—just a single mono mic.

Last but not least, throw up a large-diaphragm condenser like a Neumann U 87 to record the natural ambience. If you're really flush, record the room in stereo, or position one room mic close and one farther away. I like to pan the tracks from an audience perspective (hi-hat right, floor tom left). Keep track of which overhead is which so that you don't accidentally reverse the panning and smear the whole drum image. The same goes for the room mics. If you have any leftover mics, ask the drummer if he wants to sing.

David Darlington's latest projects are the Wayne Shorter CD Alegria *(Verve), Russell Gunn's* Ethnomusicology Volume 3 *(Justin Time), and the theme for the Levinson/Fontana TV series "The Joke Show."*

For audio use, the difference between a fast computer today and a fast computer from two years ago is dramatic.

PERCEPTION AND REALITY

BIGGER AND ... YES, BETTER!

I JUST BUILT A COMPUTER, AND FOR THE FIRST time, I wasn't knocked out by how much faster it was than the one it replaced. One of the really fun things about being a computer geek and having the skill to upgrade and build machines on the cheap is that I usually have top-quality machines around the house. I keep on top of what's going on in the hardware world and usually build a new machine once a year, or if it's not worth that hassle, I'll bump up a few components to add some juice.

A fairly major Windows XP problem left me thinking seriously about installing the operating system again on my audio computer, which was due for some updates. The chipset hadn't been tolerant of some audio cards (not a good thing for someone who reviews them). It was using only 256 MB of obsolete SDRAM. I guessed that I would upgrade in the next six months anyway, so if I was going to install XP and all of my audio applications again, why not do that only once? I kept the same hard drives, CD burner, graphics card, and DVD-ROM (all more than adequate); bumped up to an Nforce2 motherboard, an Athlon 2400+, and a gigabyte of very high-quality (and expensive), low-latency Corsair memory; and picked up a new case. Installing XP took about twice as long as it took to assemble the hardware, and installing my audio applications took about five times as long as both of the former put together.

Finally, after hours of dropping install CDs into the disc tray, chasing down serial numbers and registration cards, finding the most recent drivers, and generally fiddling with stuff, my computer was ready. I broke a bottle of champagne over its prow, dubbed it the Upsetter, and prepared for a little whiplash from its searing 0–60 performance.
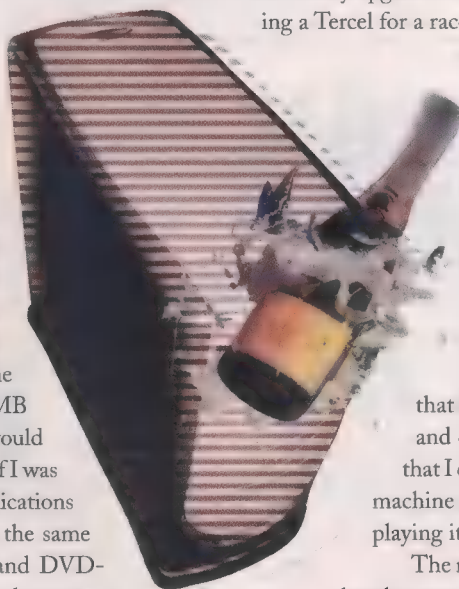
It was not to be. Windows booted a bit more quickly—maybe a few seconds faster. Applications opened in about the same amount of time as on earlier machines. General Windows tasks, file searches and copies, and software installations felt a bit snappier. But with every upgrade to previous machines, I had felt like I was trading a Tercel for a race car, while this felt like trading a Tercel for a Camry. Luckily, though, I was just asking the machine to show off with the wrong tasks. When I finally started working with audio applications, the change in performance was impossible to miss. I could load plug-ins like crazy, I got lots more voices on soft synths before the computer started to stress, and it was flat-out impossible to get Reason to complain. I loaded up a combination Cubase/Reason song that had seriously maxed my previous computer and caused me to render tracks to audio files so that I could get some CPU bandwidth, and the new machine had number-crunching horsepower to spare playing it back.

The moral of the story? Processors are now so fast that they are no longer a primary bottleneck in day-to-day computer performance. When I write, Web surf, and answer e-mail, my 733 MHz Pentium III isn't noticeably different than the Athlon 2400+. For audio use, though, the difference between a fast computer today and a fast computer from two years ago is dramatic, which is a good thing for musicians and audio computer users. Just don't let yourself be fooled by the performance in non-audio tasks. That's not where new hardware will shine.

LEVIN PFEUFER

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While reverb is one of the most common effects, it can also have a high degree of nuance and complexity.

CAUSE AND EFFECTS

SIGNAL PROCESSING BASICS, PART 1: REVERB

SIGNAL PROCESSING IS A CORE COMPONENT of modern recording. You can record a stellar performance, but it is often the skilled use of effects processing that takes your material to the next level. Today's technology has put quality digital signal processing within reach of the most modest home recording studio, as the effects created by the technology have become an integral part of modern music. As sound design wizards continue to invent new ways to twist, bend, and mangle audio, it is often the creative use of effects that distinguishes one professional recording from another in a popular market that thrives on new techno thrills.

So you finally have your hands on that new magic box or plug-in that is going to give your tracks the edge, but now what? You can just reach for the presets and hope for the best, but before long, you are going to want to fine-tune the effects for your needs. Suddenly, you're faced with a sea of parameters, and just blindly tweaking settings won't cut it. To unlock the magic, you'll need an understanding of the basic concepts behind today's digital effects and how their editing parameters function and interrelate. Over the next several Home Clinics, we'll be taking a look at the basics of signal processing and the power behind the presets, starting with the most widely used studio effect of all, reverberation.

Reverb is the result of reflected sound within an enclosed space. It can add excitement and drama to dry music tracks, or lend a sense of space and environment to film and TV postproduction. While reverb is one of the most common effects, it can also have a high degree of nuance and complexity. Over the years, engineers have developed a range of techniques for generating the sound of both natural and artificial reverberant spaces. Digital re-creations of these classic sounds are available as presets in today's reverb processing devices. The choices and labels may vary slightly from one device to another, but they generally fall into the following categories.

Halls. Hall programs, designed to re-create the sound of a concert hall, typically come in small, medium, and large sizes. They tend to be the richest and smoothest programs and are great for vocals, pianos, and orchestral instruments. The very largest halls may have extremely long decays, like that of a gothic cathedral.

Chambers. Reverberation chambers were specially designed rooms used in early recording studios. They were fairly large rooms with hard surfaces, in which loudspeakers were played and picked up by microphones. Chamber programs tend to sound smooth and similar to halls, but with more mids and highs and a more even tonal color as the sound decays. They are useful for a wide range of material and especially good for spoken voice applications.

Plate Reverbs. Prior to the digital age, plate reverbs were used on many classic recordings. Large sheets of metal with attached transducers vibrate within an enclosed frame to simulate natural reverb. Made by companies like EMT, plate units are still maintained by a few top studios and create reverbs that are bright, smooth, dense, and relatively colored.

Rooms/Ambience. These types of programs emulate the sound of various live spaces—bathrooms, conference rooms, marble foyers, etc.—and are reflective without overly coloring the direct sound.

Specialized Reverbs. The gated reverb presents an initial explosive reverberant sound that dies away abruptly. It can add dramatic effect, but because of its quick decay, it won't drown the mix. It's typically used on transient sounds like drums or percussion. Reverse reverb is a special effect that runs the reverb algorithm backward to create a swell that sharply closes off into a "*shhooop*" effect.

Next time, we'll take a look at the various editing parameters of digital reverbs, what they mean, and how to use them.

Babz is a freelance writer on music technology and a composer and multi-instrumentalist based in New York City.

LEVN PFEUFER

By David Simons

Sticky Fingers was an instant classic, a diverse collection that included the band's first two Top 40 hits of the '70s.

THE ROLLING STONES
STICKY FINGERS
A Little Brown Sugar at the Shoals

IN THE FALL OF 1969, THE ROLLING STONES' MICK Jagger and Keith Richards were suddenly hitting a creative high. *Let It Bleed* was in the can, and already "the glimmer twins" had new material ready to be studio tested. After concluding a brief U.S. tour in West Palm Beach (featuring new guitarist Mick Taylor) on November 30, the band decided to check out a new facility in Sheffield, Alabama, dubbed Muscle Shoals Sound Studios.

When the Stones rolled into town on December 2, Muscle Shoals Sound—a former casket factory conveniently located next to a graveyard—was still a fledgling operation, with only a modicum of pro gear and just one hit to date (R.B. Greaves' smash "Take a Letter

Maria"). Recording the Stones, however, was "the big time," as former session player and co-owner Barry Beckett later put it. After giving the Scully 8-track a quick once-over, Beckett's partner Jimmy Johnson, the studio's self-appointed engineer, positioned himself in the MSS control room and patiently waited for "the world's greatest rock 'n' roll band" to arrive.

"The Stones came in, and they were a little rusty at first, because they hadn't been practicing on account of the tour," says Johnson. "It took them some time to get in the groove. On everything they did, they'd be going in all different directions for about two or three hours. So I just took it upon myself to have the tape machine ready and get rolling as soon as they started getting good."

The studio's 10-input Universal Audio console dictated a frugal miking scheme, with three mics assigned to Charlie Watts' drum kit and one SM57 each on Richards' and Taylor's Fender amps. With the exception of Richards' amp (which was isolated due to loudness), leakage was encouraged; even Jagger's lead vocals were tracked live, using a Neumann U 47. "I also put a 47 over Charlie's drum kit," says Johnson. "That, combined with those UA pre's and the open room, produced the nicest drum sound. Charlie definitely agreed!" So did Richards. "There was something about the Muscle Shoals feel," remarked the guitarist years later. "Charlie really filled the sound—and it was so easy to cut down there."

On their first night in the Shoals, the Stones delivered a faithful cover of Mississippi Fred McDowell's Delta blues classic "You Gotta Move." On December 3 came "Brown Sugar," a glorious Chuck Berry update written by Jagger. The final evening produced yet another classic in "Wild Horses," Richards' paean to country pal Gram Parsons (which utilized Richards' newest love, an acoustic guitar in Nashville high-string tuning).

Despite the hours of studio noodling, once the band gelled, the takes came together quickly, recalls Johnson. "And all of a sudden, Keith would come up with a figure, and the band would just follow in around it. By hour five or six, it had really started to come together. In the meantime, they wouldn't be saying anything to me, but I knew I had to get the very best performance when it happened. After a few takes of 'Wild Horses,' Jagger just looks up at me and says, 'Is that it?'—like I'm the producer or something! But I knew when they had it—and I just told them to come out and hear it back."

Although it's believed that the noticeably distorted vocal on "Wild Horses" was Jagger's deliberate attempt to achieve a swampy southern sound, "it was the UA console that did that," Johnson says with a grin. "At the time, we had an EV monitor speaker that had a real distortion problem in the amp. The thing is, Mick kept harping

on it, but I finally convinced him that what he was hearing really wasn't going onto the tape. Ironically, there was this other natural distortion caused by those old mic preamps that really was going onto the tape! In the end, it added to the overall ambience and gave the song real character."

Additional cuts "Sway," "Moonlight Mile" (both without Richards), and "Bitch" were completed at Jagger's English estate in early 1970, using the group's brand-new mobile recording unit. Yet it took the band until April 1971—nearly a year and a half after the

Album notes
Sticky Fingers

RELEASED
April 23, 1971
PRODUCER
Jimmy Miller
ENGINEERS
Jimmy Johnson, Glyn Johns, Andy Johns, Chris Kimsey
RECORDED AT
Olympic Sound Studios, London;

Stargroves (Jagger's home), Newbury, UK; Muscle Shoals Sound Studios, Sheffield, Alabama, December 1969–April 1970
HIGHEST POSITION
#1
SALES
Certified gold

Alabama getaway—to finally deliver their new product. *Sticky Fingers* was an instant classic, a diverse collection that included the band's first two Top 40 hits of the '70s: "Brown Sugar" (#1) and "Wild Horses" (#28). Setting the pace for the many outlandish acts to follow was *Sticky Fingers'* audacious cover art: a pelvic-region photograph of a man in form-fitting jeans photographed by artist Andy Warhol (and including an actual working zipper). Though the rumors persisted for years, it can now be told: The bulge isn't Mick's. ▌▐▌▐

Dave Simons is a senior contributor to Home Recording. *He recently programmed all the gear in his multitrack studio to only reproduce classic rock.*

By Arty Skye

Trust Your Ears
They Know More Than Others Think

When I work at different studios with different equipment, I always bring one special tool along for a reference point. I'm talking about my ears. I've learned that what your ears tell you is much more valuable than what you might measure on a piece of test equipment.

One morning in the middle of an album project, I brought up a mix from the previous night's session. Something sounded different. I had trouble putting my finger on it, but I knew something was wrong. The assistant swore that nobody had been in the studio, and he ran a recall of the computer-based automation system, which came up perfect. Nobody else heard a difference, and the assistant and tech were looking at me strangely. But I trust my ears, and I wasn't going to continue the mix until we corrected the error.

They reexamined all the patches and settings, and everything checked out. Then they put the alignment tape up on the slave 2" machine, and we found the culprit: Channels 9–16 were down about 1 dB. The maintenance staff eventually figured out that a part in the power supply for that middle section had blown, lowering the output level on those tracks. They replaced the part, and we were back in business.

I trust my ears, and they almost never let me down. Being cognizant of the way your ear perceives sound is an integral part of achieving a good mix. Three phenomena known as *masking*, *beating*, and *combination tones* play significant roles.

When you have many different sounds competing for the same frequencies, they begin to cover up one another. This is called *masking*. For example, if you have an acoustic guitar that sounds great with a 3–5 kHz boost, and you're boosting those same frequencies on the lead vocals, percussion, and strings, you will probably get a wash in that area of the spectrum, with nothing really standing out. That's one reason not to solo instruments when equalizing. It is not the isolated sounds but how the sounds relate to one another and sit in the track that makes the difference between a lousy mix and a great one. Masking also occurs when louder sounds cover up softer ones.
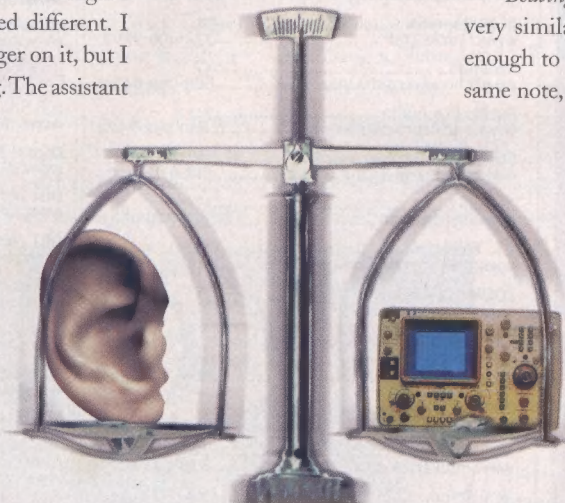
Beating occurs when you have two very similar frequencies that are close enough to each other to be basically the same note, but different enough to cause a beating sound when played simultaneously. Think of when you're tuning a guitar to a specific note from a keyboard or a tone (not a guitar tuner). The beating will sound like a fast vibrato, but it will slow down and eventually disappear as you approach the proper tuning.

Combination tones are tones that are perceived to be audible but don't really occur. We hear these tones as the difference between two frequencies if they differ by more than 50 Hz. If one instrument is playing a 3 kHz tone and another is playing a 2.5 kHz tone, we will perceive a phantom tone occurring at 500 Hz, or the difference between the two (3 kHz − 2.5 kHz = 500 Hz).

If you're hearing any of these things while you're mixing but can't seem to find them by soloing the instruments, start soloing in pairs and then threes until you can isolate and eradicate the problem.

Your ears are incredible tools capable of extremely fine sound analysis. The best part is that they go everywhere you do. Learn to trust them. ▮▮▮▮

Producer/engineer Arty Skye has worked with artists such as Madonna, Brandy, Will Smith, and Santana. He owns SkyeLab Studio (skyelab.com) and realengineers.com.

LEVIN PFEUFER